Voice over IP for the Cisco AS5300

Feature Summary

Voice over IP (VoIP) enables a Cisco AS5300 access server to carry voice traffic (for example, telephone calls and faxes) over an IP network. VoIP is primarily a software feature; however, to use this feature on the Cisco AS5300, you must install a VoIP feature card (VFC). Each VFC can hold up to five digital signal processor modules (DSPMs). The VFC utilizes the Cisco AS5300's quad T1/E1 Public Switched Telephone Network (PSTN) interface and LAN or WAN routing capabilities to provide up to a 48/60 channel gateway for VoIP packetized voice traffic. For more information about the physical characteristics, installing, or configuring a VFC in your Cisco AS5300 access server, refer to *Installing Voice over IP Feature Cards in Cisco AS5300 Universal Access Servers*, which came with your your VFC.

VoIP for the Cisco AS5300 has two primary applications:

- It provides a central-site telephony termination facility for VoIP traffic from multiple voice-equipped remote office facilities.
- It provides a PSTN gateway for Internet telephone traffic. VoIP used as a PSTN gateway leverages the standardized use of H.323-based Internet telephone client applications.

Figure 1 and Figure 2 illustrate these applications.

Figure 1 VoIP Used as a Central-Site Telephony Termination Facility

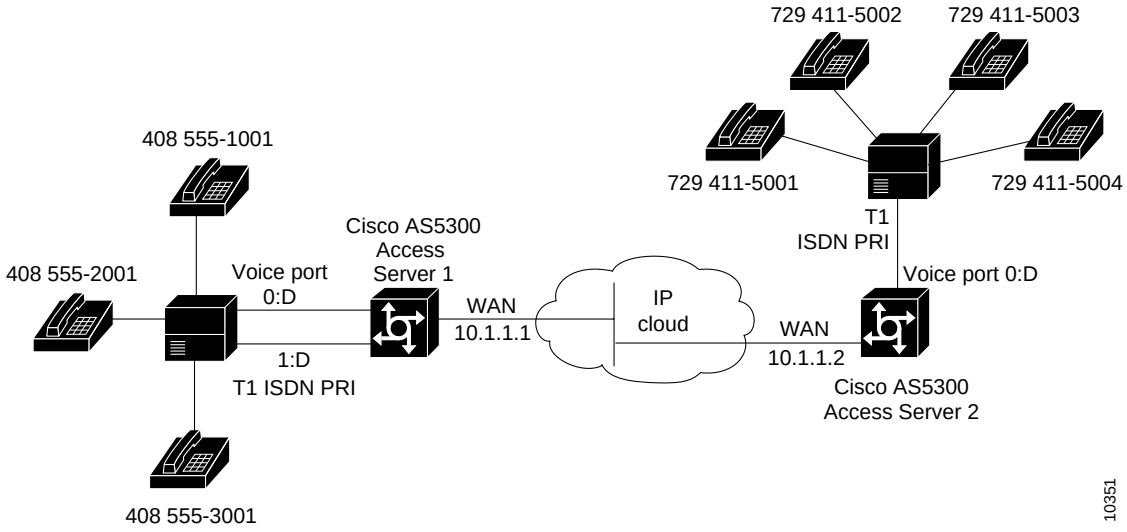
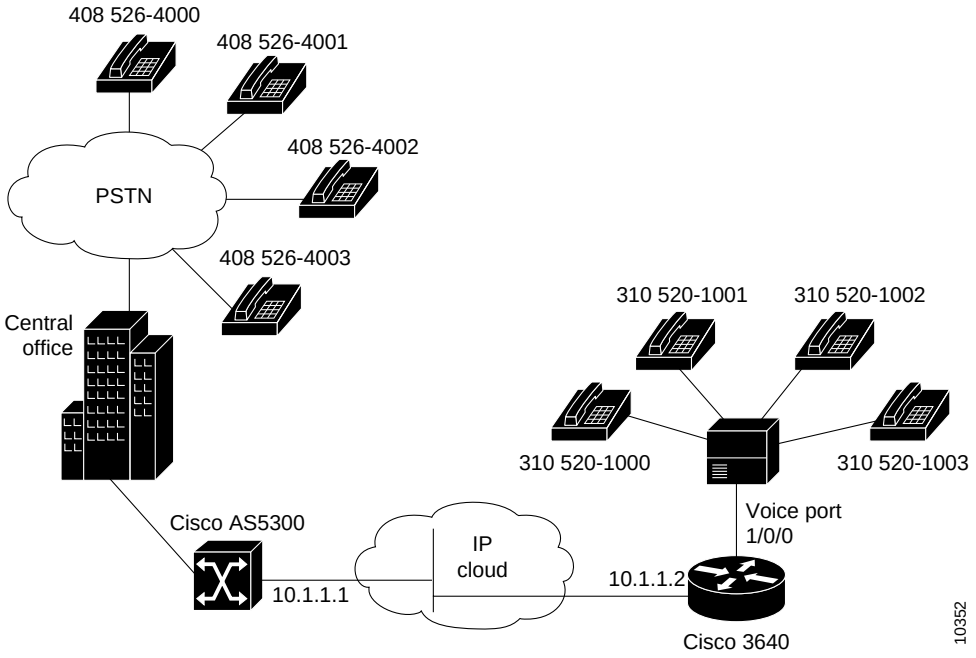


Figure 2 VoIP Used as a PSTN Gateway for Internet Telephone Traffic



How VoIP Processes a Telephone Call

Before configuring VoIP on your Cisco AS5300, it helps to understand what happens at an application level when you place a call using VoIP. The general flow of a two-party voice call using VoIP is as follows:

1. The user picks up the handset; this signals an off-hook condition to the signalling application part of VoIP in the Cisco AS5300.
2. The session application part of VoIP issues a dial tone and waits for the user to dial a telephone number.
3. The user dials the telephone number; those numbers are accumulated and stored by the session application.
4. After enough digits are accumulated to match a configured destination pattern, the telephone number is mapped to an IP host via the dial plan mapper. The IP host has a direct connection to either the destination telephone number or a PBX that is responsible for completing the call to the configured destination pattern.
5. The session application then runs the H.323 session protocol to establish a transmission and a reception channel for each direction over the IP network. If the call is being handled by a PBX, the PBX forwards the call to the destination telephone. If RSVP has been configured, the RSVP reservations are put into effect to achieve the desired quality of service (QoS) over the IP network.
6. The CODECs are enabled for both ends of the connection and the conversation proceeds using RTP/UDP/IP as the protocol stack.

7. Any call-progress indications (or other signals that can be carried in-band) are cut through the voice path as soon as an end-to-end audio channel is established. Signalling that can be detected by the voice ports (for example, in-band DTMF digits after the call setup is complete) is also trapped by the session application at either end of the connection and carried over the IP network encapsulated in RTCP using the RTCP APP extension mechanism.
8. When either end of the call hangs up, the RSVP reservations are torn down (if RSVP is used) and the session ends. Each end becomes idle, waiting for the next off-hook condition to trigger another call setup.

Benefits

- Toll bypass
- Remote PBX presence over WANs
- Unified voice/data trunking
- POTS-Internet telephony gateways

List of Terms

ACOM—Term used in G.165, "General Characteristics of International Telephone Connections and International Telephone Circuits: Echo Cancellers." ACOM is the combined loss achieved by the echo canceller, which is the sum of the echo return loss, echo return loss enhancement, and nonlinear processing loss for the call.

A-law—A companding technique commonly used in Europe. A-law is standardized as a 64-kbps CODEC in ITU-T G.711.

Call leg—A logical connection between the router and either a telephony endpoint over a bearer channel, or another endpoint using a session protocol.

CAS—Channel associated signalling. In E1 applications, timeslot 16 is used to transmit CAS information. Each frame's timeslot 16 carries signalling information (ABCD bits) for two of the B-channel timeslots.

CIR—Committed information rate. The average rate of information transfer a subscriber (for example, the network administrator) has stipulated for a Frame Relay PVC.

CODEC—coder-decoder. Device that typically uses pulse code modulation to transform analog signals into a digital bit stream and digital signals back into analog signals. In Voice over IP, it specifies the voice coder rate of speech for a dial peer.

Data link connection identifier (DLCI)—Frame Relay virtual circuit number corresponding to a particular destination. The DLCI is part of the Frame Relay header and is usually 10 bits long.

Dial peer—An addressable call endpoint. In Voice over IP, there are two kinds of dial peers: POTS and VoIP. In Voice over IP, you use dial peers to assign particular characteristics to call legs.

DS0—A 64-kbps channel on an E1 or T1 WAN interface.

DSP—Digital Signal Processor.

DTMF—Dual tone multifrequency. Use of two simultaneous voice-band tones for dial (such as touch tone).

E1—Wide-area digital transmission scheme. E1 is the European equivalent of a T1 line. The E1's higher clock rate (2.048 MHz) allows for 32 64-kbps channels, which include one channel for framing and one channel for D-channel information.

FIFO—First-in, first-out. In data communication, FIFO refers to a buffering scheme where the first byte of data entering the buffer is the first byte retrieved by the CPU. In telephony, FIFO refers to a queueing scheme where the first calls received are the first calls processed.

ISDN—Integrated Services Digital Network. ISDN is a communications protocol, offered by telephone companies, that permits telephone networks to carry data, voice, and other traffic.

Multilink PPP—Multilink Point-to-Point Protocol. This protocol is a method of splitting, recombining, and sequencing datagrams across multiple logical data links.

PBX—Private Branch Exchange. Privately owned central switching office.

PLAR—Private Line Auto Ringdown. PLAR is a leased voice circuit that connects two telephones. When either telephone handset is lifted, the other telephone automatically rings.

POTS—Plain old telephone service. Basic telephone service supplying standard single-line telephones, telephone lines, and access to the Public Switched Telephone Network.

POTS dial peer—Dial peer connected via a traditional telephony network. POTS peers point to a particular voice port on a voice network device.

PRI—Primary Rate Interface. PRI is an ISDN interface to primary rate access. Primary rate access consists of a single 64-kbps D channel plus 23 T1 or 30 E1 B channels for voice or data.

PSTN—Public Switched Telephone Network. PSTN refers to the local telephone company.

PVC—Permanent virtual circuit.

QoS—Quality of service, which refers to the measure of service quality provided to the user.

RSVP—Resource Reservation Protocol. This protocol supports the reservation of resources across an IP network.

T1—Digital WAN carrier facility. T1 transmits DS1 formatted data at 1.544 Mbps through the telephone-switching network, using AMI or B8ZS coding. T1 is the North American equivalent of an E1 line.

Trunk—Service that allows quasi-transparent connections between two PBXs, a PBX and a local extension, or some ther combination of telephony interfaces to be permanently conferenced together by the esession application and signalling passed transparently through the IP network.

U-law—A companding technique commonly used in North America. U-law is standardized as a 64-kbps CODEC in ITU-T G.711.

VoIP dial peer—Dial peer connected via a packet network; in the case of Voice over IP, this is an IP network. VoIP peers point to specific VoIP devices.

Platforms

The Voice over IP feature is supported on the following Cisco device platforms:

- Cisco AS5300 access servers
- Cisco 3600 series routers

The configuration procedure described in this document pertains to the Cisco AS5300. For information on how to configure Voice over IP on Cisco 3600 series routers, refer to the Cisco IOS Release 12.0 *Voice, Video, and Home Applications Configuration Guide*.

Prerequisites

Before you can configure your Cisco AS5300 to use Voice over IP, you must first do the following:

- Establish a working IP network. For more information about configuring IP, refer to the "IP Overview," "Configuring IP Addressing," and "Configuring IP Services" chapters in the Cisco IOS 12.0 *Network Protocols Configuration Guide, Part 1*.
- Complete basic configuration for the AS5300, which includes, as a minimum, the following tasks:
  - Configure a host name and password for the AS5300
  - Configure the Ethernet 10BaseT/100BaseT interface of your AS5300 so that it can be recognized as a device on the Ethernet LAN
  - Configure the AS5300 interfaces for ISDN PRI lines
  - Configure the ISDN D channels for each ISDN PRI line

  For more information about any of the these configuration tasks, refer to the *Cisco AS5300 Universal Access Server Software Configuration Guide*.
- Install the VFC into the appropriate slot of your Cisco AS5300 access server. Each VFC can hold up to five digital signal processor modules (DSPMs), enabling processing for up to 30 B channels. For more information about the physical characteristics of the VFCs or DSPMs, or how to install them, refer to *Installing Voice over IP Feature Cards in Cisco AS5300 Universal Access Servers*, which came with your VFC.
- Complete your company's dial plan.
- Establish a working telephony network based on your company's dial plan.
- Integrate your dial plan and telephony network into your existing IP network topology. Merging your IP and telephony networks depends on your particular IP and telephony network topology. In general, Cisco recommends the following suggestions:
  - Use canonical numbers wherever possible. It is important to avoid situations where numbering systems are significantly different on different routers or access servers in your network.
  - Make routing and dialing transparent to the user—for example, avoid secondary dial tones from secondary switches, where possible.
  - Contact your PBX vendor for instructions about how to reconfigure the appropriate PBX interfaces.

Supported MIBs and RFCs

This feature supports the following MIBs:

- CISCO-ANALOG-VOICE-IF-MIB
- CISCO-VOICE-DIAL-CONTROL-MIB
- CISCO-VOICE-IF-MIB

For descriptions of supported MIBs and how to use MIBs, see Cisco's MIB Web site on CCO at http://www.cisco.com/public/sw-center/netmgmt/cmtk/mibs.shtml.

This feature supports the following RFCs:

- RFC 1889—*RTP: A Transport Protocol for Real-Time Applications*, January 1996; H. Schulzrinne, GMD Fokus; S. Casner, Precept Software, Inc; R. Frederick, Xerox Palo Alto Research Centre; V. Jacobson, Lawrence Berkeley National Laboratory
- RFC 1890—*RTP Profile for Audio and Video Conferences with Minimal Control*, January 1996; H. Schulzrinne, GMD Fokus
- RFC 2127—*ISDN Management Information Base using SMIv2*, March 1997; G. Roeck, Editor; Cisco Systems
- RFC 2128—*Dial Control Management Information Base using SMIv2*, March 1997; G. Roeck, Editor; Cisco Systems
- ITU-T H.323—*Packet-Based Multimedia Communications Systems*, February 1998
- ITU-T Q.400-490 series—*Signalling System R2*, 1988 to 1993

Configuration Tasks

After you have analyzed your dial plan and decided how to integrate it into your existing IP network, you are ready to configure your network devices to support Voice over IP. The actual configuration procedure depends entirely on the topology of your voice network, but in general you need to perform the following tasks:

- Configure IP Networks for Real-Time Voice Traffic
  - Configure Multilink PPP with Interleaving
  - Configure RTP Header Compression
  - Configure Custom Queueing
  - Configure Weighted Fair Queueing
- Configure Frame Relay for Voice Over IP (if needed for your network topology)
- Configure Voice Ports
  - Configure ISDN PRI Voice Ports
  - Configure E1 R2 Voice Ports
  - Configure T1 CAS Voice Ports
- Configure Number Expansion
  - Create a Number Expansion Table
  - Configure Number Expansion
- Configure Dial Peers
  - Create a Peer Configuration Table
  - Configure POTS Peers
  - Configure VoIP Peers

Depending on the topology of your network or the resources used in your network, you might need to perform the following additional tasks:

- Distinguish Voice and Modem Calls on the Cisco AS5300
- Optimize Dial Peer and Network Interface Configurations
  - Configure IP Precedence for Dial Peers
  - Configure RSVP for Dial Peers
  - Configure CODEC and VAD for Dial Peers
- Configure Voice over IP for Microsoft NetMeeting

Voice over IP for the Cisco AS5300 also offers VFC management features that enable you to easily upgrade and manage the system software stored in VFC Flash memory. You might need to perform the following tasks to manage VCWare or DSPWare:

- Download VCWare
- Copy Flash Files to the VFC
  - Download VCWare to the VFC from the AS5300 Motherboard
  - Download VCWare to the VFC from a TFTP Server
- Unbundle VCWare
- Add Files to the Default File List
- Add CODECs to the Capability List
- Delete Files from VFC Flash Memory
- Erase the VFC Flash Memory

All of these tasks are described in the following sections.

Configure IP Networks for Real-Time Voice Traffic

You need to have a well-engineered network end-to-end when running delay-sensitive applications such as VoIP. Fine-tuning your network to adequately support VoIP involves a series of protocols and features geared toward quality of service (QoS). It is beyond the scope of this document to explain the specific details relating to wide-scale QoS deployment. Cisco IOS software provides many tools for enabling QoS on your backbone, such as Random Early Detection (RED), Weighted Random Early Detection (WRED), fancy queueing (meaning custom, priority, or weighted fair queueing), and IP Precedence. To configure your IP network for real-time voice traffic, you need to consider the entire scope of your network, then select the appropriate QoS tool or tools.

It is important to remember that QoS must be configured throughout your network—not just on the AS5300 devices running VoIP—to improve voice network performance. Not all QoS techniques are appropriate for all network routers. Edge routers and backbone routers in your network do not necessarily perform the same operations; the QoS tasks they perform might differ as well. To configure your IP network for real-time voice traffic, you need to consider the functions of both edge and backbone routers in your network, then select the appropriate QoS tool or tools.

In general, edge routers perform the following QoS functions:

- Packet classification
- Admission control
- Bandwidth management

- Queueing

In general, backbone routers perform the following QoS functions:

- High-speed switching and transport
- Congestion management
- Queue management

Scalable QoS solutions require cooperative edge and backbone functions.

Note In a subsequent Cisco IOS release, we have implemented enhancements to improve QoS on low speed, wide-area links, such as ISDN, MLPPP, and Frame Relay running on edge routers. For more information about these enhancements, refer to the Cisco IOS Release 12.0(5)T "IP RTP" feature module.

Although they are not mandatory, some QoS tools have been identified as being valuable in fine-tuning your network to support real-time voice traffic. To configure your IP network for QoS using these tools, perform one or more of the following tasks:

- Configure Multilink PPP with Interleaving
- Configure RTP Header Compression
- Configure Custom Queueing
- Configure Weighted Fair Queueing

Each of these components is discussed in the following sections.

Configure Multilink PPP with Interleaving

Multiclass Multilink PPP interleaving allows large packets to be multilink-encapsulated and fragmented into smaller packets to satisfy the delay requirements of real-time voice traffic; small real-time packets, which are not multilink-encapsulated, are transmitted between fragments of the large packets. The interleaving feature also provides a special transmit queue for the smaller, delay-sensitive packets, enabling them to be transmitted earlier than other flows. Interleaving provides the delay bounds for delay-sensitive voice packets on a slow link that is used for other best-effort traffic.

Note Interleaving applies only to interfaces that can configure a multilink bundle interface. These interfaces include virtual templates, dialer interfaces, and Integrated Services Digital Network (ISDN) Basic Rate Interface (BRI) or Primary Rate Interface (PRI) interfaces.

In general, Multilink PPP with interleaving is used in conjunction with weighted fair queueing and RSVP or IP Precedence to ensure voice packet delivery. Use Multilink PPP with interleaving and weighted fair queueing to define how data will be managed; use RSVP or IP Precedence to give priority to voice packets.

You should configure Multilink PPP if the following conditions exist in your network:

- Point-to-point connection using PPP encapsulation
- Slow links

Note Multilink PPP should not be used on links greater than 2 Mbps.

Multilink PPP support for interleaving can be configured on virtual templates, dialer interfaces, and ISDN BRI or PRI interfaces. To configure interleaving, you need to perform the following tasks:

- Configure the dialer interface or virtual template, as defined in the relevant chapters of the Cisco IOS Release 12.0 *Dial Solutions Configuration Guide*.
- Configure Multilink PPP and interleaving on the interface or template.

Enable Multilink PPP and Interleaving

To configure Multilink PPP and interleaving on a configured and operational interface or virtual interface template, use the following commands in interface configuration mode:

Step	Command	Purpose
1	**ppp multilink**	Enables Multilink PPP.
2	**ppp multilink interleave**	Enables real-time packet interleaving.
3	**ppp multilink fragment-delay** *milliseconds*	Optionally, configures a maximum fragment delay.
4	**ip rtp reserve** *lowest-UDP-port range-of-ports* [*maximum-bandwidth*]	Reserves a special queue for real-time packet flows to specified destination User Datagram Protocol (UDP) ports, allowing real-time traffic to have higher priority than other flows. This command applies only if you have not configured RSVP.

Note The **ip rtp reserve** command can be used instead of configuring RSVP. If you configure RSVP, this command is not required.

For more information about Multilink PPP, refer to the the Cisco IOS Release 12.0 *Dial Solutions Configuration Guide*.

Multilink PPP Configuration Example

The following example defines a virtual interface template that enables Multilink PPP with interleaving and a maximum real-time traffic delay of 20 milliseconds, and then applies that virtual template to the Multilink PPP bundle:

```
interface virtual-template 1
 ppp multilink
 encapsulated ppp
 ppp multilink interleave
 ppp multilink fragment-delay 20
 ip rtp reserve 16384 100 64

multilink virtual-template 1
```

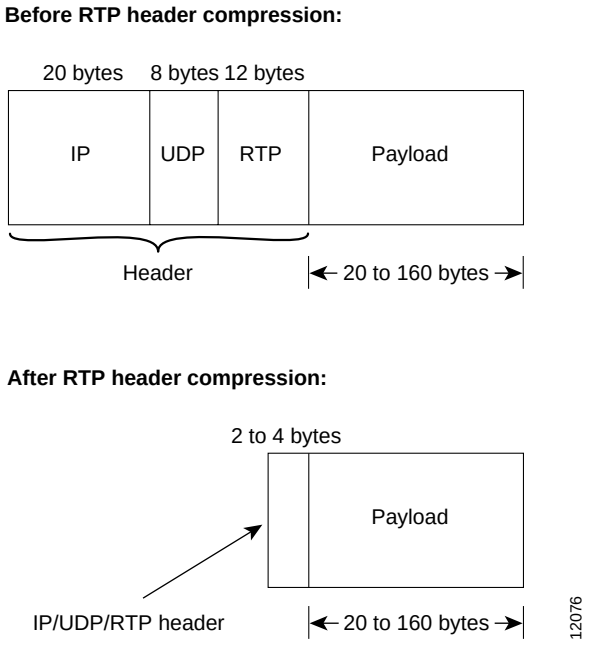
Configure RTP Header Compression

Real-Time Transport Protocol (RTP) is used for carrying packetized audio traffic over an IP network. RTP header compression compresses the IP/UDP/RTP header in an RTP data packet from 40 bytes to approximately 2 to 4 bytes (most of the time), as shown in Figure 3.

This compression feature is beneficial if you are running Voice over IP over slow links. Enabling compression on both ends of a low-bandwidth serial link can greatly reduce the network overhead if there is substantial RTP traffic on that slow link.

Typically, an RTP packet has a payload of approximately 20 to 160 bytes for audio applications that use compressed payloads. RTP header compression is especially beneficial when the RTP payload size is small (for example, compressed audio payloads of 20 to 50 bytes).

Figure 3 RTP Header Compression



You should configure RTP header compression if the following conditions exist in your network:

- Slow links
- Need to save bandwidth

Note RTP header compression should not be used on links greater than 2 Mbps.

Perform the following tasks to configure RTP header compression for Voice over IP. The first task is required; the second task is optional.

- Enable RTP Header Compression on a Serial Interface
- Change the Number of Header Compression Connections

Enable RTP Header Compression on a Serial Interface

To use RTP header compression, you need to enable compression on both ends of a serial connection. To enable RTP header compression, use the following command in interface configuration mode:

Command	Purpose
ip rtp header-compression [**passive**]	Enables RTP header compression.

If you include the **passive** keyword, the software compresses outgoing RTP packets only if incoming RTP packets on the same interface are compressed. If you use the command without the **passive** keyword, the software compresses all RTP traffic.

Change the Number of Header Compression Connections

By default, the software supports a total of 16 RTP header compression connections on an interface. To specify a different number of RTP header compression connections, use the following command in interface configuration mode:

Command	Purpose
ip rtp compression connections *number*	Specifies the total number of RTP header compression connections supported on an interface.

RTP Header Compression Configuration Example

The following example enables RTP header compression for a serial interface:

```
interface 0:23
 ip rtp header-compression
 encapsulation ppp
 ip rtp compression-connections 25
```

For more information about RTP header compression, see the Cisco IOS Release 12.0 *Network Protocols Configuration Guide, Part 1*.

Configure Custom Queueing

Some QoS features, such as IP RTP reserve and custom queueing, are based on the transport protocol and the associated port number. Real-time voice traffic is carried on UDP ports in the range 16384 to 16624. This number is derived from the following formula:

```
16384 = 4(number of voice ports in the AS5300)
```

Custom queueing and other methods for identifying high priority streams should be configured for these port ranges. For more information about custom queueing, refer to the the Cisco IOS Release 12.0 *Quality of Service Solutions Configuration Guide*.

Configure Weighted Fair Queueing

Weighted fair queueing ensures that queues do not starve for bandwidth and that traffic gets predictable service. Low-volume traffic streams receive preferential service; high-volume traffic streams share the remaining capacity, obtaining equal or proportional bandwidth.

In general, weighted fair queueing is used in conjunction with Multilink PPP with interleaving and RSVP or IP Precedence to ensure that voice packet delivery. Use weighted fair queueing with Multilink PPP to define how data will be managed; use RSVP or IP Precedence to give priority to voice packets. For more information about weighted fair queueing, refer to the Cisco IOS Release 12.0 *Quality of Service Solutions Configuration Guide*.

Configure Frame Relay for Voice Over IP

You need to consider certain factors when configuring Voice over IP for it to run smoothly over Frame Relay. A public Frame Relay cloud provides no guarantees for QoS. For real-time traffic to be transmitted in a timely manner, the data rate must not exceed the CIR or there is the possibility that packets will be dropped. In addition, Frame Relay traffic shaping and RSVP are mutually exclusive, which is particularly important to remember if multiple DLCIs are carried on a single interface.

For Frame Relay links with slow output rates (less than or equal to 64 kbps), where data and voice are being transmitted over the same PVC, Cisco recommends the following solutions:

- Separate DLCIs for voice and data—By providing a separate subinterface for voice and data, you can use the appropriate QoS tool per line. For example, each DLCI would use 32 kbps of a 64-kbps line.
  - Apply adaptive traffic shaping to both DLCIs.
  - Use RSVP or IP Precedence to prioritize voice traffic.
  - Use compressed RTP to minimize voice packet size.
  - Use weighted fair queueing to manage voice traffic.
- Lower MTU size—Voice packets are generally small. If you lower the MTU size (for example, to 300 bytes), large data packets can be broken up into smaller data packets that can more easily be interwoven with voice packets.

Note Some applications do not support a smaller MTU size. If you decide to lower MTU size, use the **ip mtu** command; this command affects only IP traffic.

Note Lowering the MTU size affects data throughput speed.

- CIR equal to line rate—Make sure that the data rate does not exceed the CIR by using generic traffic shaping.
  - Use compressed RTP to minimize voice packet header size.
- Traffic shaping—Use adaptive traffic shaping to throttle back the output rate based on the backward explicit congestion notification (BECN) bit. If the feedback from the switch is ignored, packets (both data and voice) might be discarded. Because the Frame Relay switch does not distinguish between voice and data packets, voice packets could be discarded, which would result in a deterioration of voice quality.
  - Use compressed RTP, reduced MTU size, and adaptive traffic shaping based on BECN to hold data rate to CIR.
  - Use generic traffic shaping to obtain a low interpacket wait time. For example, set the Bc parameter to 4000 to obtain an interpacket wait of 125 milliseconds.

Note We recommend FRF.12 fragmentation setup rules for Voice over IP connections over Frame Relay. FRF.12 was implemented in the Cisco IOS Release 12.0(4)T. For more information, refer to the Cisco IOS Release 12.0(4)T "Voice over Frame Relay using FRF.11 and FRF.12" feature module.

Frame Relay for Voice over IP Configuration Example

For Frame Relay, it is customary to configure a main interface and several subinterfaces, one subinterface per PVC. The following example configures a Frame Relay main interface and a subinterface so that voice and data traffic can be successfully transported:

```
interface Serial0/0
 ip mtu 300
 no ip address
 encapsulation frame-relay
 no ip route-cache
 no ip mroute-cache
 fair-queue 64 256 1000
 frame-relay ip rtp header-compression

interface Serial0/0.1 point-to-point
 ip mtu 300
 ip address 40.0.0.7 255.0.0.0
 ip rsvp bandwidth 48 48
 no ip route-cache
 no ip mroute-cache
 bandwidth 64
 traffic-shape rate 32000 4000 4000
 frame-relay interface-dlci 16
 frame-relay ip rtp header-compression
```

In this configuration example, the main interface has been configured as follows:

- MTU size of IP packets is 300 bytes.
- No IP address is associated with this serial interface. The IP address must be assigned for the subinterface.
- Encapsulation method is Frame Relay.
- Fair queueing is enabled.
- IP RTP header compression is enabled.

The subinterface has been configured as follows:

- MTU size is inherited from the main interface.
- IP address for the subinterface is specified.
- Bandwidth is set to 64 kbps.
- Generic traffic shaping is enabled with 32-kbps CIR where Bc = 4000 bits and Be = 4000 bits.
- Frame Relay DLCI number is specified.
- IP RTP header compression is enabled.

Note When traffic bursts over the CIR, output rate is held at the speed configured for the CIR (for example, traffic will not go beyond 32 kbps if CIR is set to 32 kbps).

For more information about Frame Relay, refer to the Cisco IOS Release 12.0 *Wide-Area Networking Configuration Guide*.

Configure Voice Ports

When an interface on the Cisco AS5300 is carrying voice data, it is referred to as a voice port. Voice over IP on the Cisco AS5300 is supported over three different interface types in this release:

- ISDN PRI
- E1R2 Signalling
- T1-CAS Signalling

Note A voice port was created automatically when you installed the VFC in the Cisco AS5300 and configured an ISDN PRI group. Configuring an ISDN PRI group is part of the basic Cisco AS5300 configuration procedure. For more information, refer to the *Cisco AS5300 Universal Access Server Software Configuration Guide*.

Configure ISDN PRI Voice Ports

With ISDN PRI, signalling in Voice over IP for the AS5300 is handled by ISDN PRI group configuration. After ISDN PRI has been configured for both B and D channels for both ISDN PRI lines, you need to issue the **isdn incoming-voice** command on the serial interface (acting as the D channel) to ensure a dial tone.

Under most circumstances, the default voice-port command values are adequate to configure voice ports to transport voice data over your existing IP network. Because of the inherent complexities involved with PBX networks, you might need specific voice-port values configured, depending on the specifications of the devices in your telephony network. For more information regarding specific voice-port configuration commands, refer to the “Command Reference” section of this document.

Configure ISDN PRI for Voice over IP

To configure a voice port, use the following commands beginning in global configuration mode:

Step	Command	Purpose
1	**isdn switch-type** *switch-type*	Defines the telephone company’s switch type.
2	**controller T1 0**	Enables the T1 0 controller and enters controller configuration mode.
3	**framing esf**	Defines the framing characteristics.
4	**clock source line primary**	Configures one T1 line to serve as the primary clock source.
5	**linecode** *value*	Sets the line code type to match that of your telephone company service provider.
6	**pri-group timeslots** *range*	Configures ISDN PRI.
7	**controller T1 1**	Enables the T1 1 controller and enters controller configuration mode.
8	**framing esf**	Defines the framing characteristics.
9	**linecode** *value*	Sets the line code type to match that of your telephone company service provider.

Step	Command	Purpose
10	**pri-group timeslots** *range*	Configures ISDN PRI.
11	**interface Serial0:23**	Configures the IDSN D channel for the first ISDN PRI line. (The serial interface is the D channel.)
12	**ip address** *ip-address*	Specifies an IP address for the interface.
13	**isdn incoming-voice** {**voice** | **modem**}	Enables incoming ISDN voice calls.
14	**interface Serial1:23**	Configures the IDSN D channel for the second ISDN PRI line.
15	**ip address** *ip-address*	Specifies an IP address for the interface.
16	**isdn incoming-voice** {**voice** | **modem**}	Enables incoming ISDN voice calls.

Verify ISDN PRI Configuration

You can check the validity of your voice port configuration by performing the following tasks:

- Use the **show voice port** command to verify that the data configured is correct.
- If you have not configured your device to support direct inward dial (DID), dial in to the router and see if you have dial tone.
- Enter a DTMF digit. If the dial tone stops, you have two-way voice connectivity with the router.

Tips

If you are having trouble connecting a call and you suspect the problem is associated with voice-port configuration, you can try to resolve the problem by performing the following tasks:

- Ping the associated IP address to confirm connectivity. If you cannot successfully ping your destination, refer to the "Configuring IP" chapter in the Cisco IOS 12.0 *Network Protocols Configuration Guide, Part 1*.
- Determine if the VFC has been correctly installed. For more information, refer to *Installing Voice-over-IP Feature Cards in Cisco AS5300 Universal Access Servers*, which came with your voice network module (VNM).
- Use the **show vfc slot number** command to learn if the VFC is operational.
- Use the **show isdn status** command to view layer status information. If you receive a status message stating that Layer 1 is deactivated, make sure the cable connection is not loose or disconnected. (This status message indicates a problem at the physical layer.)
- With T1 lines, determine if your a-law setting is correct. With E1 lines, determine if your u-law setting is correct. Use the **cptone** command to configure both a-law or u-law values. For more information about the **cptone** command, refer to the "Command Reference" section of this document.
- If dialing cannot occur, use the **debug isdn q931** command to check the ISDN configuration.

Configure E1 R2 Voice Ports

The Voice over IP VNM for the Cisco AS5300 supports E1 R2 signalling as well as ISDN PRI. R2 signalling is an international signalling standard that is common to channelized E1 networks. However, there is no single signalling standard for R2. The ITU-T Q.400-Q.490 recommendation

defines R2, but a number of countries and geographic regions implement R2 in entirely different ways. Cisco Systems addresses this lack of standards by supporting many localized implementations of R2 signalling in its Cisco IOS software.

Cisco Systems' E1 R2 signalling default is ITU, which supports the technology used in the following countries: Denmark, Finland, Germany, Russia (ITU variant), Hong Kong (ITU variant), and South Africa (ITU variant). The expression "ITU variant" means there are multiple R2 signalling types in the specified country, but Cisco supports the ITU variant.

Cisco Systems also supports specific local variants of E1 R2 signalling in the following regions, countries, and corporations:

- Argentina
- Australia
- Brazil
- China
- Colombia
- Costa Rica
- East Europe (includes Croatia, Russia, and the Slovak Republic)
- Ecuador ITU
- Ecuador LME
- Greece
- Guatemala
- Hong Kong (China variant)
- Indonesia
- Israel
- Korea
- Malaysia
- Mexico (Telmex corporation)
- Mexico (Telnor corporation)
- New Zealand
- Paraguay
- Peru
- Philippines
- Saudi Arabia
- Singapore
- South Africa (Panaftel variant)
- Thailand
- Uruguay
- Venezuela
- Vietnam

Of the local variants listed above, the following local variants have been verified:

- Argentina
- Brazil
- China
- Mexico (Telmax)
- Singapore
- Thailand

R2 signalling is channelized E1 signalling used in Europe, Asia, and South America. It is equivalent to channelized T1 signalling in North America. There are two types of R2 signalling: line signalling and interregister signalling. R2 line signalling includes R2 digital, R2 analog, and R2 pulse. R2 interregister signalling includes R2 compelled, R2 noncompelled, and R2 semicompelled. These signalling types are configured using the **cas-group** command.

Many countries and regions have their own E1 R2 variant specifications, which supplement the ITU-T Q.400-Q.490 recommendation for R2 signalling. Unique E1 R2 signalling parameters for specific countries and regions are set by entering the **cas-custom** *channel* command followed by the **country** *name* command.

Cisco's implementation of R2 signalling has dialed number identification service (DNIS) support turned on by default. If you enable the automatic number identification (**ani**) option, the collection of DNIS information is still performed. Specifying the **ani** option does not disable DNIS collection. DNIS is the number being called. ANI is the caller's number. For example, if you are configuring router A to call router B, then the DNIS number is assigned to router B; the ANI number is assigned to router A. ANI is similar to Caller ID.

Configure E1 R2 Signalling for Voice over IP

To configure E1 R2 signalling, use the following commands beginning in global configuration mode:

Step	Command	Purpose
1	**controller e1** *number*	Specifies the E1 controller that you want to configure with R2 signalling.
2	**cas-group** *channel* **timeslots** *range* **type** {**r2-analog** | **r2-digital** | **r2-pulse**} [**dtmf** | **r2-compelled** [**ani**] | **r2-non-compelled** [**ani**] | **r2-semi-compelled** [**ani**]]	Configures R2 channel-associated signalling on the E1 controller. For a complete description of the available R2 options, refer to the **cas-group (controller e1)** command in the Cisco IOS Release 12.0 *Dial Solutions Command Reference*.
3	**cas-custom** *channel*	Enters cas-custom mode. In this mode, you can localize E1 R2 signalling parameters, such as specific R2 country settings for Hong Kong. For the customization to take effect, the *channel* number used in the **cas-custom** command must match the *channel* number specified by the **cas-group** command.

Step	Command	Purpose
4	**country** *name* **use-defaults**	Specifies the local country, region, or corporation specification to use with R2 signalling. Replace the *name* variable with one of the supported country names. Cisco strongly recommends that you include the **use-defaults** option, which engages the default settings for a specific country. The default setting for all countries is ITU. See the **cas-custom** command in the Cisco IOS Release 12.0 *Dial Solutions Command Reference* for the list of supported regions, countries, or corporation specifications.
5	• **ani-digits** • **answer-signal** • **caller-digits** • **category** • **default** • **dnis-digits** • **invert-abcd** • **ka** • **kd** • **metering** • **nc-congestion** • **unused-abcd** • **request-category**	(Optional) Further customizes the R2 signalling parameters. Some switch types require you to fine tune your R2 settings. Do not tamper with these commands unless you fully understand your switch's requirements. For nearly all network scenarios, the **country** *name* **use-defaults** command fully configures your country's local settings. You should not need to perform Step 5. See the **cas-custom** command in the Cisco IOS Release 12.0 *Dial Solutions Command Reference* for more information about each signalling command.
6	**exit**	Exits interface configuration mode.
7	**voice-port** *controller-number***:***channel-number*	Enters voice-port configuration mode for the specified voice port.
8	**cptone** *country-code*	Defines the country-specific PCM encoding and tones. The PCM encoding type must match the country code defined by the **cas-custom** command.
9	**exit**	Exits voice-port configuration mode.
10	**exit**	Exits global configuration mode.

As mentioned in the previous configuration steps, the E1 R2 signalling type (whether ITU, ITU variant, or local variant as defined by the **cas-custom** command) needs to match the appropriate PCM encoding type as defined by the **cptone** command. For countries for which a **cptone** value has not yet been defined, you can try the following:

- If the country uses a-law E1 R2 signalling, use the **GB** value for the **cptone** command.
- If the country uses u-law E1 R2 signalling, use the **US** value for the **cptone** command.

For more information about configuring R2 signalling, refer to the Cisco IOS Release 12.0 *Dial Solutions Configuration Guide*.

Verify E1 R2 Signalling Configuration

To verify the E1 R2 signalling configuration:

- Type the **show controller e1** command to view the status for all controllers, or type the **show controller e1** *number* command to view the status for a particular controller. Make sure the status indicates the controller is up (line 2 in the following example) and no alarms (line 4 in the following example) or errors (lines 9 and 10 in the following example) have been reported.

```
5300# show controller e1 0

E1 0 is up.
  Applique type is Channelized E1 - balanced
  No alarms detected.
  Version info of Slot 0:  HW: 2, Firmware: 4, PLD Rev: 2

Manufacture Cookie is not programmed.

  Framing is CRC4, Line Code is HDB3, Clock Source is Line Primary.
  Data in current interval (785 seconds elapsed):
     0 Line Code Violations, 0 Path Code Violations
     0 Slip Secs, 0 Fr Loss Secs, 0 Line Err Secs, 0 Degraded Mins
     0 Errored Secs, 0 Bursty Err Secs, 0 Severely Err Secs, 0 Unavail Secs
  Total Data (last 13 15 minute intervals):
     0 Line Code Violations, 0 Path Code Violations,
     0 Slip Secs, 12 Fr Loss Secs, 0 Line Err Secs, 0 Degraded Mins,
     0 Errored Secs, 0 Bursty Err Secs, 0 Severely Err Secs, 12 Unavail Secs
```

- To check the robbed-bit signalling status of each channel, type the **debug serial interface** command and the **show controller e1** command.

```
as5300#debug serial interface

Serial network interface debugging is on

as5300#show controller e1 0

E1 0 is up.
  Applique type is Channelized E1 - balanced
  No alarms detected.
  Version info of Slot 0: HW:2, Firmware:4, PLD Rev:0

Manufacture Cookie Info:
 EEPROM Type 0x0001, EEPROM Version 0x01, Board ID 0x43,
 Board Hardware Version 1.0, Item Number 73-2218-4,
 Board Revision A0, Serial Number 07805788,
 PLD/ISP Version 0.0, Manufacture Date 19-Feb-1998.

  Framing is NO-CRC4, Line Code is HDB3, Clock Source is Line Primary.
  Data in current interval (135 seconds elapsed):
     0 Line Code Violations, 0 Path Code Violations
     0 Slip Secs, 0 Fr Loss Secs, 0 Line Err Secs, 0 Degraded Mins
     0 Errored Secs, 0 Bursty Err Secs, 0 Severely Err Secs, 0 Unavail
Secs
 Robbed bit signals state:
        timeslots       rxA rxB rxC rxD         txA txB txC txD

        1               0   0   0   1           0   1   0   1
        2               0   0   0   1           0   1   0   1
        3               0   0   0   1           0   1   0   1
        4               1   0   0   1           1   0   0   1
        5               1   0   0   1           1   0   0   1
        6               0   0   0   1           0   1   0   1
        7               1   0   0   1           1   0   0   1
```

```
8               1   0   0   1           1   0   0   1
9               1   0   0   1           1   0   0   1
10              1   0   0   1           1   0   0   1
11              0   0   0   1           0   1   0   1
12              0   0   0   0           0   0   0   0
13              1   0   0   1           1   0   0   1
14              1   0   0   1           1   0   0   1
15              1   0   0   1           1   0   0   1
17              0   0   0   1           0   1   0   1
18              1   0   0   1           1   0   0   1
19              1   0   0   1           1   0   0   1
20              0   0   0   1           0   1   0   1
21              1   0   0   1           1   0   0   1
22              1   0   0   1           1   0   0   1
23              1   0   0   1           1   0   0   1
24              0   0   0   1           0   1   0   1
25              0   0   0   1           0   1   0   1
26              0   0   0   1           0   1   0   1
27              0   0   0   1           0   1   0   1
28              0   0   0   0           0   0   0   0
29              0   1   1   0           0   1   0   1
30              0   1   0   1           0   0   0   1
31              0   0   0   0           1   0   0   0
```

Tips

If the connection does not come up, check for the following:

- Loose wires, splices, connectors, shorts, bridge taps, and grounds
- Backward transmit and receive
- Mismatched framing types (for example, CRC-4 versus no-CRC-4)
- Transmit and receive pair separation (crosstalk)
- Faulty line cards or repeaters
- Noisy lines (for example, power and crosstalk)

If you see errors on the line or the line is going up and down, check for the following:

- Mismatched line codes—for example, high density bipolar 3 (HDB3) versus alternate mark inversion (AMI)
- Receive level
- Frame slips due to poor clocking plan

Configure T1 CAS Voice Ports

CAS is the transmission of signalling information within the voice channel. Various types of CAS signalling are available in the T1 world. The most common forms of CAS signalling are loop-start, ground-start, and E&M. The main disadvantage of CAS signalling is its use of user bandwidth to perform signalling functions. CAS signalling is often referred to as robbed-bit signalling because user bandwidth is being “robbed” by the network for other purposes. In addition to receiving and placing calls, CAS signalling processes the receipt of DNIS and ANI information, which is used to support authentication and other functions.

T1 CAS capabilities have been implemented on the Cisco AS5300 VFC to enhance and integrate T1 CAS capabilities on common central office (CO) and PBX configurations for voice calls. The service provider application for T1 CAS includes connectivity to the public network using T1 CAS from the Cisco AS5300 to the end office switch. In this configuration, the Cisco AS5300 captures the dialed-number or called-party number information and passes it along to the upper level applications for interactive voice response (IVR) script selection, modem pooling, and other applications. Service providers also require access to calling party number, ANI, for user identification, for billing account number, and in the future, for more complicated call routing.

Service providers who implement VoIP include traditional voice carriers, new voice and data carriers, and existing Internet service providers. Some of these service providers might use subscriber side lines for their VoIP connectivity to the PSTN; others might use tandem-type service provider connections.

T1 CAS Signalling Systems

Voice over IP for the AS5300 supports the following T1 CAS signalling systems:

- E&M—E&M signalling is typically used for trunks. It is normally the only way that a CO switch can provide two-way dialing with direct inward dialing. In all the E&M protocols, off-hook is indicated by A = B = 1, and on-hook is indicated by A = B = 0. If dial pulse dialing is used, the A and B bits are pulsed to indicate the addressing digits. There are several further important subclasses of E&M robbed-bit signalling:
  - E&M Wink Start—Feature Group B

    In the original Wink Start protocol, the terminating side responds to an off-hook from the originating side with a short wink (transition from on-hook to off-hook and back again). This wink tells the originating side that the terminating side is ready to receive addressing digits. After receiving addressing digits, the terminating side then goes off-hook for the duration of the call. The originating endpoint maintains off-hook for the duration of the call.
  - E&M Wink Start—Feature Group D

    In Feature Group D Wink Start with Wink Acknowledge protocol, the terminating side responds to an off-hook from the originating side with a short wink (transition from on-hook to off-hook and back again) just as in the original Wink Start. This wink tells the originating side that the terminating side is ready to receive addressing digits. After receiving addressing digits, the terminating side then provides another wink (called an Acknowledgment Wink) that tells the originating side that the terminating side has received the dialed digits. The terminating side then goes off-hook to indicate connection when the ultimate called endpoint has answered. The originating endpoint maintains off-hook for the duration of the call.

— E&M Immediate Start

In the Immediate Start protocol, the originating side does not wait for a wink before sending addressing information. After receiving addressing digits, the terminating side then goes off-hook for the duration of the call. The originating endpoint maintains off-hook for the duration of the call.

- Ground Start / FXS—Ground Start signalling was developed to aid in resolving glare when two sides of a connection tried to go off-hook at the same time. Two sides of the connection simultaneously going off-hook creates a problem with loop start signalling because the only way an incoming call from the network was recognized by the customer premise equipment (CPE) using loop start was to ring the phone. The 6-second ring cycle left a substantial amount of time for glare to occur. Ground Start signalling eliminates this problem by providing an immediate seizure indication from the network to the CPE. This indication tells the CPE that a particular channel has an incoming call on it. Ground Start is different than E&M in that the A and B bits do not track each other (that is, A is not necessarily equal to B). When the CO delivers a call, it "seizes" a channel (goes off-hook) by setting the A bit to 0. The CO equipment also simulates ringing by toggling the B bit. The terminating equipment goes off-hook when it is ready to answer the call. Digits are usually not delivered for incoming calls.

Channelized T1 Robbed-Bit Features

Internet service providers can provide switched 56-kbps access to their customers using the Cisco AS5300. The subset of T1 CAS (robbed bit) supported features are as follows:

Supervisory: Line Side

- fxs-loop-start
- fxs-ground-start
- sas-loop-start
- sas-ground-start
- Modified R1

Supervisory: Trunk Side

- e&m-fgb
- e&m-fgd
- e&m-immediate-start

Informational: Line Side

- DTMF

Informational: Trunk Side

- DTMF
- MF

Configure T1 CAS for Voice over IP

To configure T1 CAS for Voice over IP on the Cisco AS5300, use the following commands beginning in privileged EXEC mode:

Step	Command	Purpose
1	**configure terminal**	Enters global configuration mode.
2	**controller t1** *number*	Enters controller configuration mode to configure your controller port. The controller ports are labeled 0 through 3 on the Quad T1/PRI and E1/PRI cards.
3	**framing** {**sf** | **esf**}	Enters the framing type designated by your telephone company.
4	**clock source line primary**	Configures the primary PRI clock source. Configure other lines as secondary or internal clock sources. Note that only one PRI can be clock source primary and one PRI can be clock source secondary.
5	**linecode** {**ami** | **b8zs** | **hdb3**}	Enters the line code type designated by your telephone company.
6	**cas-group** *channel* **timeslots** *range* **type** *signal*	Configures all channels for E&M, FXS, and SAS analog signalling. Enter **1-24** for T1. If E1, type **1-31**. Signalling types include **e&m-fgb**, **e&m-fgd**, **e&m-immediate-start**, **fxs-ground-start**, **fxs-loop-start**, **sas-ground-start**, and **sas-loop-start**. You must use the same type of signalling that your central office uses. For E1 using the Anadigicom converter, use **cas e&m-fgb** signalling.
7	**controller t1** *number*	Enters controller configuration mode to configure the second controller port (There are a total of four controller ports). The controller ports are labeled 0 through 3 on the Quad T1/PRI and E1/PRI cards.
8	**framing** {**sf** | **esf**}	Enters the framing type designated by your telephone company.
9	**clock source line secondary**	Configures the secondary PRI clock source. Note that only one PRI can be clock source primary and one PRI can be clock source secondary.
10	**linecode** {**ami** | **b8zs** | **hdb3**}	Enters the line code type designated by your telephone company.

Step	Command	Purpose
11	**cas-group** *channel* **timeslots** *range* **type** *signal*	Configures all channels for E&M, FXS, and SAS analog signalling. Enter **1-24** for T1. If E1, enter **1-31**. Signalling types include **e&m-fgb**, **e&m-fgd**, **e&m-immediate-start**, **fxs-ground-start**, **fxs-loop-start**, **sas-ground-start**, and **sas-loop-start**. You must use the same type of signalling that your central office uses. For E1 using the Anadigicom converter, use **cas e&m-fgb** signalling.
12	**controller t1** *number*	Enters controller configuration mode to configure the third controller port (there are a total of four controller ports). The controller ports are labeled 0 through 3 on the Quad T1/PRI and E1/PRI cards.
13	**framing** {**sf** | **esf**}	Enters the framing type designated by your telephone company.
14	**clock source line internal**	Configures the internal PRI clock source. Note that only one PRI can be clock source primary and one PRI can be clock source secondary. All other controller ports use an internal PRI clock source.
15	**linecode** {**ami** | **b8zs** | **hdb3**}	Enters the line code type designated by your telephone company.
16	**cas-group** *channel* **timeslots** *range* **type** *signal*	Configures all channels for E&M, FXS, and SAS analog signalling. Type **1-24** for T1. If E1, type **1-31**. Signalling types include **e&m-fgb**, **e&m-fgd**, **e&m-immediate-start**, **fxs-ground-start**, **fxs-loop-start**, **sas-ground-start**, and **sas-loop-start**. You must use the same type of signalling that your central office uses. For E1 using the Anadigicom converter, use **cas e&m-fgb** signalling. Repeat steps 12 through 16 to configure the last controller.

Verify T1 CAS Configuration

To verify your controller is up and running and no alarms have been reported, perform the following task:

- Enter the **show controller t1** or **show controller e1** command and specify the port number.

```
5300# show controller t1 2
 T1 2 is up.
   No alarms detected.
   Version info of slot 0:  HW: 2, Firmware: 16, PLD Rev: 0

 Manufacture Cookie Info:
  EEPROM Type 0x0001, EEPROM Version 0x01, Board ID 0x42,
  Board Hardware Version 1.0, Item Number 73-2217-4,
  Board Revision A0, Serial Number 06467665,
  PLD/ISP Version 0.0, Manufacture Date 14-Nov-1997.

   Framing is ESF, Line Code is B8ZS, Clock Source is Internal.
   Data in current interval (269 seconds elapsed):
    0 Line Code Violations, 0 Path Code Violations
      0 Slip Secs, 0 Fr Loss Secs, 0 Line Err Secs, 0 Degraded Mins
      0 Errored Secs, 0 Bursty Err Secs, 0 Severely Err Secs, 0 Unavail Secs
```

Note the following:

— The controller must report being up.

— No errors should be reported.

Tip

Make sure the **show controller t1** output is not reporting alarms or violations.

Configure Number Expansion

In most corporate environments, the telephone network is configured so that you can reach a destination by dialing only a portion (an extension number) of the full E.164 telephone number. Voice over IP can be configured to recognize extension numbers and expand them into their full E.164 dialed number by using two commands in tandem: **destination-pattern** and **num-exp**. Before you configure these two commands, it is helpful to map individual telephone extensions with their full E.164 dialed numbers. This mapping can be done easily by creating a number expansion table.

Create a Number Expansion Table

In Figure 4, a small company wants to use Voice over IP to integrate its telephony network with its existing IP network. The destination pattern (or expanded telephone number) associated with Access Server 1 (located to the left of the IP cloud) is (408) 555-xxxx, where xxxx identifies the individual dial peers by extension. The destination pattern (or expanded telephone number) associated with Access Server 2 (located to the right of the IP cloud) is (729) 411-xxxx.

Figure 4 Sample Voice over IP Network

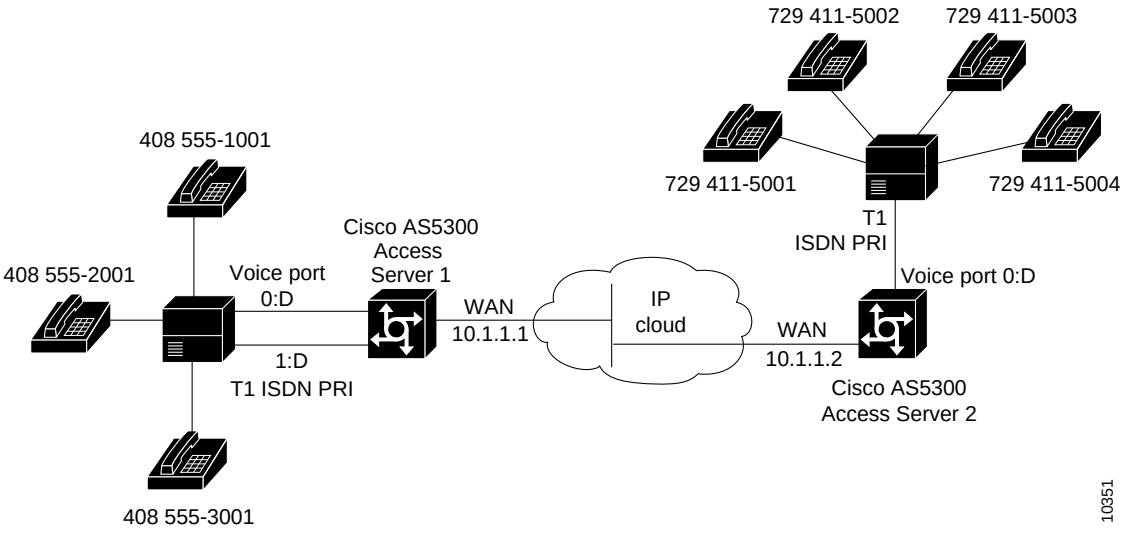


Table 1 shows the number expansion table for this scenario.

Table 1 Sample Number Expansion Table

Extension	Destination Pattern	Num-Exp Command Entry
1...	408555....	num-exp 1... 408555....
2...	408555....	num-exp 2... 408555....
3...	408555....	num-exp 3... 408555....
4...	7294115...	num-exp 4.... 7294115...

Note You can use the period symbol (.) to represent variables (such as extension numbers) in a telephone number.

The information included in this example needs to be configured on both Router 1 and Router 2.

Configure Number Expansion

To define how to expand an extension number into a particular destination pattern, use the following command in global configuration mode:

Command	Purpose
num-exp *extension-number extension-string*	Configures number expansion.

You can verify the number expansion information by using the **show num-exp** command to display the telephone number mapping.

After you have configured dial peers and assigned destination patterns to them, you can verify number expansion information by using the **show dialplan number** command to learn how a telephone number maps to a dial peer.

Configure Dial Peers

The key point to understanding how Voice over IP functions is to understand dial peers. Each dial peer defines the characteristics associated with a call leg, as shown in Figure 5 and Figure 6. Dial peers are used to apply attributes to call legs and to identify call origin and destination. Attributes applied to a call leg include QoS, CODEC, VAD, and fax rate. A call leg is a discrete segment of a call connection that lies between two points in the connection. All of the call legs for a particular connection have the same connection ID.

An end-to-end call is comprised of four call legs, two from the perspective of the source router or access server as shown in Figure 5, and two from the perspective of the destination router or access server as shown in Figure 6. A dial peer is associated with each one of these call legs.

Figure 5 Dial Peer Call Legs from the Perspective of the Source Router or Access Server

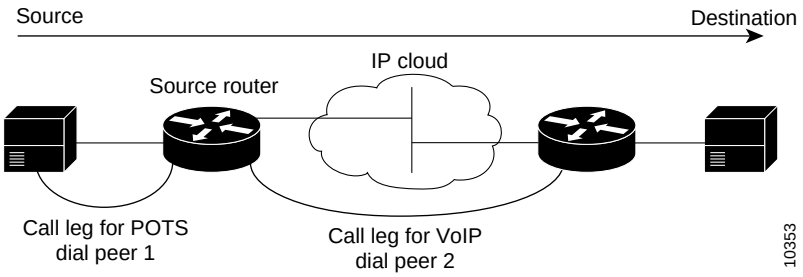
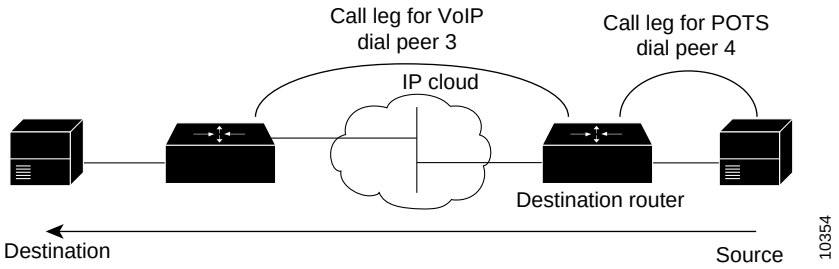


Figure 6 Dial Peer Call Legs from the Perspective of the Destination Router or Access Server



There are two different kinds of dial peers as shown in both Figure 5 and Figure 6:

POTS—POTS dial peers describe the line characteristics usually associated with a traditional telephony network; in VoIP for the Cisco AS5300, they describe the the specific line characteristics between the telephony device and the Cisco AS5300. POTS dial peers point to a particular voice port on a network device—in the case of VoIP for the Cisco AS5300, they point to a specific voice port on the Cisco AS5300 through which voice traffic will travel to the rest of the voice network.

VoIP—VoIP dial peers describe the line characteristics usually associated with a packet network connection (in the case of VoIP, this is an IP network). VoIP peers define the line characteristics between VoIP devices—the routers and access servers carrying voice traffic in this voice network.

Inbound versus Outbound Dial Peers

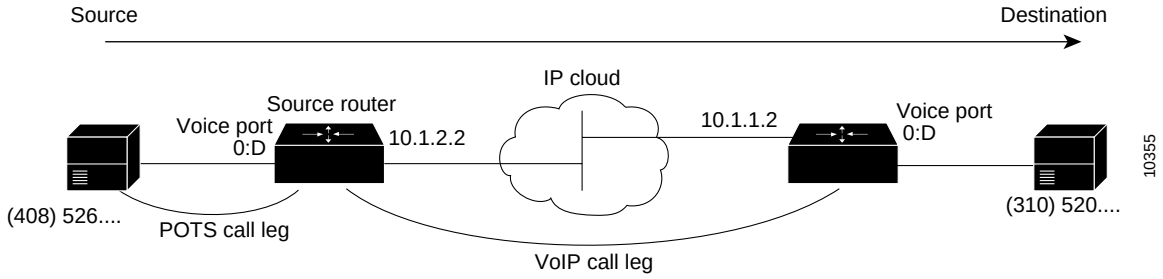
Dial peers are used for both inbound and outbound call legs. It is important to remember that these terms are defined from the access server's perspective. An inbound call leg originates *outside* the access server. An outbound call leg originates *from* the access server.

For inbound call legs, a dial peer might be associated to the calling number or the port designation. Outbound call legs always have a dial peer associated with them. The destination pattern is used to identify the outbound dial peer. The call is associated with the outbound dial peer at setup time.

POTS peers associate a telephone number with a particular voice port so that incoming calls for that telephone number can be received and outgoing calls can be placed. VoIP peers point to specific devices (by associating destination telephone numbers with a specific IP address) so that incoming calls can be received and outgoing calls can be placed. Both POTS and VoIP peers are needed to establish Voice over IP connections.

Establishing communication using Voice over IP is similar to configuring an IP static route: you are establishing a specific voice connection between two defined endpoints. As shown in Figure 7, for outgoing calls (from the perspective of the POTS dial peer 1), the POTS dial peer establishes the source (via the originating telephone number or voice port) of the call. The VoIP dial peer establishes the destination by associating the destination phone number with a specific IP address.

Figure 7 Outgoing Calls from the Perspective of POTS Dial Peer 1



To configure call connectivity between the source and destination as illustrated in Figure 7, enter the following commands on router 10.1.2.2:

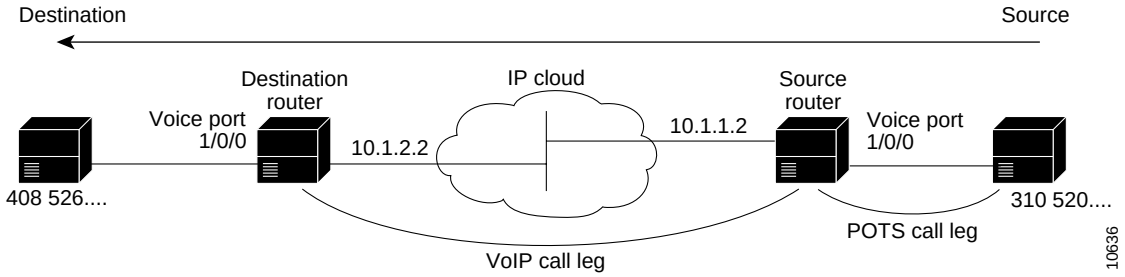
```
dial-peer voice 1 pots
 destination-pattern 1408526....
 port 0:D

dial-peer voice 2 voip
 destination-pattern 1310520....
 session target ipv4:10.1.1.2
```

In the previous configuration example, the last four digits in the VoIP dial peer's destination pattern were replaced with wildcards, which means that from router 10.1.2.2, calling any number string that begins with the digits "1310520" plus four digits will result in a connection to router 10.1.1.2. By implication, configuring the destination pattern this way means that router 10.1.1.2 services all numbers beginning with those digits. From router 10.1.1.2, calling any number string that begins with the digits "1408526" will result in a connection to router 10.1.2.2. By implication, configuring

the destination pattern this way means that router 10.1.2.2 services all numbers beginning with those digits. For more information about stripping and adding digits, see the “Outbound Dialing on POTS Peers” section in this document.

Figure 8 shows how to complete the end-to-end call between dial peer 1 and dial peer 4.

Figure 8 Outgoing Calls from the Perspective of POTS Dial Peer 2



To complete the end-to-end call between dial peer 1 and dial peer 4 as illustrated in Figure 8, enter the following commands on router 10.1.1.2:

```
dial-peer voice 4 pots
 destination-pattern 1310520....
 port 0:D

dial-peer voice 3 voip
 destination-pattern 1408526....
 session target ipv4:10.1.2.2
```

Create a Peer Configuration Table

Specific data relative to each dial peer needs to be identified before you can configure dial peers in Voice over IP. One way to organize this data before you configure VoIP is to create a peer configuration table.

Using the example in Figure 4, Router 1, with an IP address of 10.1.1.1, connects a small sales branch office to the main office through Router 2. Three telephones in the sales branch office need to be connected to Router 1 via the sales office’s PBX. Router 2, with an IP address of 10.1.1.2, is the primary gateway to the main office; as such, it needs to be connected to the company’s PBX. Four basic telephone sets need to be connected to Router 2 via the main office’s PBX. Figure 4 shows a diagram of this small voice network.

Figure 9 Sample VoIP Network

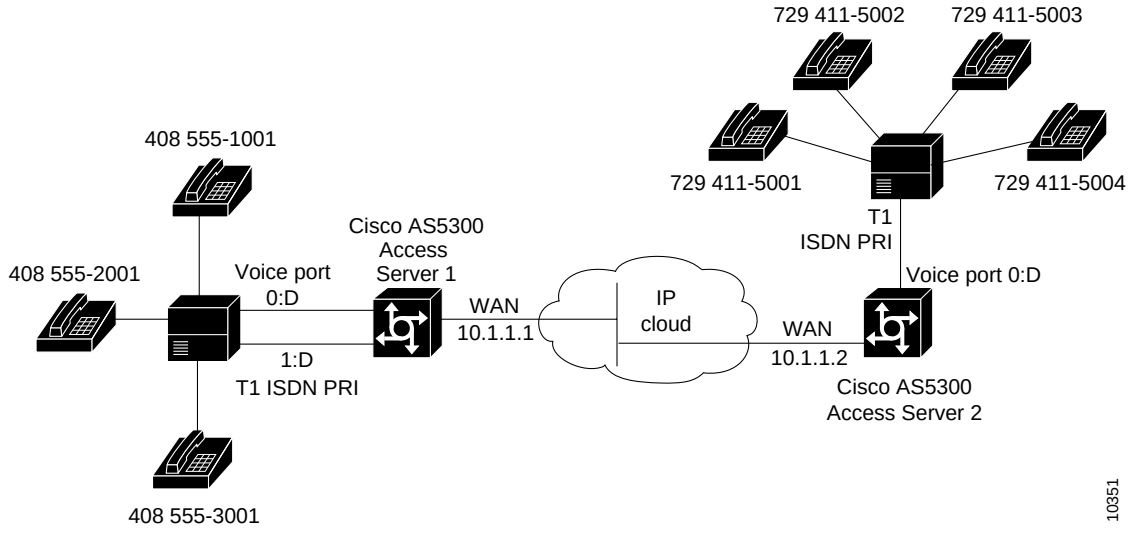


Table 2 shows the peer configuration table for the example illustrated in Figure 4.

Table 2 Peer Configuration Table for Sample Voice Over IP Network

		Commands				
Dial Peer Tag	**Ext**	**Dest-Pattern**	**Type**	**Session-Target**	**CODEC**	**QoS**
Server 1						
1	1...	+1408555....	POTS			
2	2...	+1408555....	POTS			
3	3...	+1408555....	POTS			
10		+17294115...	VoIP	IPV4 10.1.1.2	G.729	Best Effort
Server 2						
11		+1408555....	VoIP	IPV4 10.1.1.1	G.729	Best Effort
4	4...	+17294115...	POTS			

Configure POTS Peers

POTS peers enable incoming calls to be received by a particular telephony device by defining the call leg characteristics between the telephony device and the Cisco AS5300. To configure a POTS peer, you need to uniquely identify the peer (by assigning it a unique tag number), associate the peer with a voice port through which calls will be established, and define the destination telephone number(s). Under most circumstances, the default values for the remaining dial peer configuration commands will be sufficient to establish connections.

To enter the dial peer configuration mode (and select POTS as the method of voice-related encapsulation), use the following commands in the global configuration mode:

Command	Purpose
dial-peer voice *number* **pots**	Enters the dial peer configuration mode to configure a POTS peer.

The *number* value of the **dial-peer voice pots** command is a tag that uniquely identifies the dial peer. (This number has local significance only.)

To configure the identified POTS peer, use the following commands in the dial peer configuration mode:

Step	Command	Purpose
1	**destination-pattern** *string*	Defines the telephone number associated with this POTS dial peer.
2	**port** *controller number*:**D**	Associates this POTS dial peer with a specific logical dial interface.

Outbound Dialing on POTS Peers

When a router receives a voice call, it selects an outbound dial peer by comparing the called number (the full E.164 telephone number) in the call information with the number configured as the destination pattern for the POTS peer. The router then strips out the explicit left-justified numbers corresponding to the destination pattern matching the called number. If you have configured a prefix, the prefix will be prepended in front of the remaining numbers, creating a dial string, which the router will then dial. If all numbers in the destination pattern are stripped-out, the user will receive (depending on the attached equipment) a dial tone.

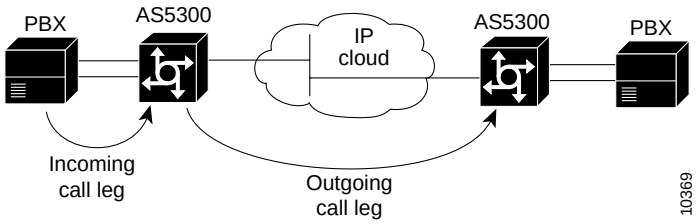
For example, suppose there is a voice call whose E.164 called number is 1(310) 555-2222. If you configure a destination pattern of “1310555” and a prefix of “9,” the router will strip out “1310555” from the E.164 telephone number, leaving the extension number of “2222.” It will then prepend the prefix “9” to the front of the remaining numbers, so that the actual numbers dialed is “9, 2222.” The comma in this example means that the router will pause for one second between dialing the “9” and the “2” to allow for a secondary dial tone.

For additional POTS dial-peer configuration options, refer to the “Command Reference” section in this document.

Direct Inward Dial for POTS Peers

Direct inward dial (DID) is used to determine how the called number is treated for incoming POTS call legs. As shown in Figure 10, incoming means from the perspective of the router. In this case, it is the call leg coming into the access server to be forwarded through to the appropriate destination pattern.

Figure 10 Incoming and Outgoing POTS Call Legs



Unless otherwise configured, when a call arrives on the access server, the server presents a dial tone to the caller and collects digits until it can identify the destination dial peer. After the dial peer has been identified, the call is forwarded through the next call leg to the destination.

There are cases where it might be necessary for the server to use the called number (DNIS) to find a dial peer for the outgoing call leg—for example, if the switch connecting the call to the server has already collected the digits. DID enables the server to match the called number with a dial peer and then directly place the outbound call. With DID, the server does not present a dial tone to the caller and does not collect digits; it forwards the call directly to the configured destination.

To use DID and incoming called-number, a dial peer must be associated with the incoming call leg. Before associating the dial peer with the incoming call leg, it helps if you understand the logic behind the algorithm used to associate the incoming call leg with the dial peer. The algorithm used to associate incoming call legs with dial peers uses three inputs (which are derived from signalling and interface information associated with the call) and four defined dial peer elements. The three signalling inputs are as follows:

- Called number (DNIS)—Set of numbers representing the destination, which is derived from the ISDN setup message or CAS DNIS.
- Calling number (ANI)—Set of numbers representing the origin, which is derived from the ISDN setup message or CAS DNIS.
- Voice port—The voice port carrying the call.

The four defined dial peer elements are as follows:

- Destination pattern—A pattern representing the phone numbers to which the peer can connect.
- Answer address—A pattern representing the phone numbers from which the peer can connect.
- Incoming called number—A pattern representing the phone numbers that associate an incoming call leg to a peer based on the called number or DNIS.
- Port—The port through which calls to this peer are placed.

Using the elements, the algorithm is as follows:

```
For all peers where call type (VoIP versus POTS) match dial peer type:
if the type is matched, associate the called number with the incoming called-number
 else if the type is matched, associate calling-number with answer-address
 else if the type is matched, associate calling-number with destination-pattern
 else if the type is matched, associate voice port to port
```

This algorithm shows that if a value is not configured for **answer-address**, the origin address is used because, in most cases, the origin address and **answer-address** are the same.

To configure DID for a particular POTS dial peer, use the following commands beginning in global configuration mode:

Step	Command	Purpose
1	**dial-peer voice** *number* **pots**	Enters the dial peer configuration mode to configure a POTS peer.
2	**direct-inward-dial**	Specifies direct inward dial for this POTS peer.

Note Direct inward dial is configured for the calling POTS dial peer.

For additional POTS dial peer configuration options, refer to the "Command Reference" section of this document.

Configure VoIP Peers

VoIP peers enable outgoing calls to be made from a particular telephony device by defining the line characteristics between the transmitting and receiving Cisco AS5300s. To configure a VoIP peer, you need to uniquely identify the peer (by assigning it a unique tag number), define its destination telephone number and destination IP address. As with POTS peers, under most circumstances, the default values for the remaining dial peer configuration commands will be adequate to establish connections.

To enter the dial-peer configuration mode (and select VoIP as the method of voice-related encapsulation), use the following command beginning in global configuration mode:

Command	Purpose
dial-peer voice *number* **voip**	Enters the dial peer configuration mode to configure a VoIP peer.

The *number* value of the **dial-peer voice voip** command is a tag that uniquely identifies the dial peer.

To configure the identified VoIP peer, use the following commands in the dial-peer configuration mode:

Step	Command	Purpose
1	**destination-pattern** *string*	Defines the destination telephone number associated with this VoIP dial peer.
2	**session-target** {**ipv4:***destination-address* | **dns:***host-name*}	Specifies a destination IP address for this dial peer.

For additional VoIP dial peer configuration options, refer to the "Commands" section of this document. For examples of how to configure dial peers, refer to the "Configuration Examples" section of this document.

Verify the Dial Peer Configuration

You can check the validity of your dial peer configuration by performing the following tasks:

- If you have relatively few dial peers configured, you can use the **show dial-peer voice** command to verify that the data configured is correct. Use this command to display a specific dial peer or to display all configured dial peers.
- Use the **show dialplan number** command to show the dial peer to which a particular number (destination pattern) resolves.

Tips

If you are having trouble connecting a call and you suspect the problem is associated with dial peer configuration, you can try to resolve the problem by performing the following tasks:

- Ping the associated IP address to confirm connectivity. If you cannot successfully ping your destination, refer to the "Configuring IP" chapter in the Cisco IOS 12.0 *Network Protocols Configuration Guide, Part 1*.
- Use the **show dial-peer voice** command to verify that the operational status of the dial peer is up.
- Use the **show dialplan number** command on the local and remote routers to verify that the data is configured correctly on both.
- If you have configured number expansion, use the **show num-exp** command to check that the partial number on the local router maps to the correct full E.164 telephone number on the remote router.
- If you have configured a CODEC value, there can be a problem if both VoIP dial peers on either side of the connection have incompatible CODEC values. Make sure that both VoIP peers have been configured with the same CODEC value.
- Use the **debug vpm spi** command to verify that the output string the router dials is correct.
- Use the **debug cch323 rtp** command to check RTP packet transport.
- Use the **debug cch323 h245** command to check logical channel negotiation.
- Use the **debug cch323 h225** command to check the call setup.

Distinguish Voice and Modem Calls on the Cisco AS5300

When the Cisco AS5300 is handling both modem and voice calls, it needs to be able to identify the service type of the call—that is, whether the incoming call to the server is a modem or a voice call. In a mixed environment, where the server receives both modem and voice calls, you need to identify the service type of the call. You can identify the service type of the call in one of two ways:

- Configure the **incoming called-number** command on the voice dial peer associated with the interface over which the call comes in
- Assign the called-number to a modem pool.

It helps to understand the logic behind the algorithm the system uses to distinguish voice and modem calls. The algorithm is as follows:

```
If the called-number matches a number from the modem pool,
 handle the call as a modem call
If the called-number matches a configured dial peer incoming called number,
 handle the call as a voice call
Else handle the call as a modem call by default modem pool
```

If there is no called number information provided within the call setup, call classification is handled as follows:

```
If there is a modem available in the system-default modem pool
 handle the call by a modem from this pool.
Else handle the call as a voice call (either ise the voice dial peer assigned to the
interface over which the call has arrived or use the default dial peer 0).
```

To identify the service type of a call to be voice, use the following commands beginning in global configuration mode:

Step	Command	Purpose
1	**dial-peer voice** *number* **pots**	Enters the dial peer configuration mode to configure a POTS peer.
2	**incoming called-number** *number*	Specifies direct inward dial for this POTS peer.

Optimize Dial Peer and Network Interface Configurations

Depending on how you have configured your network interfaces, you might need to configure additional VoIP dial peer parameters. This section describes the following topics:

- Configure IP Precedence for Dial Peers
- Configure RSVP for Dial Peers
- Configure CODEC and VAD for Dial Peers

Configure IP Precedence for Dial Peers

If you want to give real-time voice traffic a higher priority than other network traffic, you can weight the voice data traffic associated with a particular VoIP dial peer by using IP Precedence. IP Precedence scales better than RSVP but provides no admission control.

To give real-time voice traffic precedence over other IP network traffic, use the following commands beginning in global configuration mode:

Step	Command	Purpose
1	**dial-peer voice** *number* **voip**	Enter the dial peer configuration mode to configure a VoIP peer.
2	**ip precedence** *number*	Select a precedence level for the voice traffic associated with that dial peer.

In IP Precedence, the numbers 1 through 5 identify classes for IP flows; the numbers 6 and 7 are used for network and backbone routing and updates.

For example, to ensure that voice traffic associated with VoIP dial peer 103 is given a higher priority than other IP network traffic, enter the following:

```
dial-peer voice 103 voip
 ip precedence 5
```

In this example, when an IP call leg is associated with VoIP dial peer 103, all packets transmitted to the IP network via this dial peer will have their precedence bits set to 5. If the networks receiving these packets have been configured to recognize precedence bits, the packets will be given priority over packets with a lower configured precedence value.

Configure RSVP for Dial Peers

If you have configured your WAN or LAN interfaces for RSVP, you must configure the QoS for any associated VoIP peers. To configure QoS for a selected VoIP peer, use the following commands beginning in global configuration mode:

Step	Command	Purpose
1	**dial-peer voice** *number* **voip**	Enters the dial peer configuration mode to configure a VoIP peer.
2	**req-qos** [**best-effort** | **controlled-load** | **guaranteed-delay**]	Specifies the desired quality of service to be used.

Note Cisco suggests that you select **controlled-load** for the requested quality of service.

For example, to specify guaranteed delay QoS for VoIP dial peer 108, enter the following:

```
Dial-peer voice 108 voip
 destination-pattern +1408528
 req-qos controlled-load
 session target ipv4:10.0.0.8
```

In this example, every time a connection is made through VoIP dial peer 108, an RSVP reservation request is made between the local router, all intermediate routers in the path, and the final destination router.

To generate an SNMP trap message if the reserved QoS is less than the configured value for a selected VoIP peer, use the following commands beginning in the global configuration mode:

Step	Command	Purpose
1	**dial-peer voice** *number* **voip**	Enters the dial peer configuration mode to configure a VoIP peer.
2	**acc-qos** [**best-effort** | **controlled-load** | **guaranteed-delay**]	Specifies the QoS value below which an SNMP trap will be generated.

Note RSVP reservations are only one-way. If you configure RSVP, the VoIP dial peers on both ends of the connection must be configured for RSVP.

Configure CODEC and VAD for Dial Peers

Coder-decoder (CODEC) and voice activity detection (VAD) for a dial peer determine how much bandwidth the voice session uses. CODEC typically is used to transform analog signals into a digital bit stream and digital signals back into analog signals—in this case, it specifies the voice coder rate of speech for a dial peer. VAD is used to disable the transmission of silence packets.

Configure CODEC for a VoIP Dial Peer

To specify a voice coder rate for a selected VoIP peer, use the following commands beginning in global configuration mode:

Step	Command	Purpose
1	**dial-peer voice** *number* **voip**	Enters the dial peer configuration mode to configure a VoIP peer.
2	**codec** [**g711alaw** | **g711ulaw** | **g729r8**]	Specifies the desired voice coder rate of speech.

The default for the **codec** command is **g729r8**; normally the default configuration for this command is the most desirable. If, however, you are operating on a high bandwidth network and voice quality is of the highest importance, you should configure the **codec** command for G711 A Law or G711 U Law. Using either of these values will result in better voice quality, but it will also require higher bandwidth requirements for voice.

For example, to specify a CODEC rate of G.711A Law for VoIP dial peer 108, enter the following:

```
dial-peer voice 108 voip
 destination-pattern +1408528
 codec g711alaw
 session target ipv4:10.0.0.8
```

Configure VAD for a VoIP Dial Peer

To disable the transmission of silence packets for a selected VoIP peer, use the following commands beginning in global configuration mode:

Step	Command	Purpose
1	**dial-peer voice** *number* **voip**	Enters the dial peer configuration mode to configure a VoIP peer.
2	**vad**	Disables the transmission of silence packets (enabling VAD).

The default for the **vad** command is enabled; normally the default configuration for this command is the most desirable. If you are operating on a high bandwidth network and voice quality is of the highest importance, you should disable **vad**. Disabling VAD will result in better voice quality, but it will also require higher bandwidth requirements for voice.

For example, to enable VAD for VoIP dial peer 108, enter the following:

```
dial-peer voice 108 voip
 destination-pattern +1408528
 vad
 session target ipv4:10.0.0.8
```

Configure Voice over IP for Microsoft NetMeeting

Voice over IP can be used with Microsoft NetMeeting (Version 2.x) when the Cisco AS5300 is used as the voice gateway. Use the latest version of DirectX drivers from Microsoft on your PC to improve the voice quality of NetMeeting.

Configure Voice over IP to Support Microsoft NetMeeting

To configure Voice over IP to support NetMeeting, create a VoIP peer that contains the following information:

- Session Target—IP address or DNS name of the PC running NetMeeting
- CODEC—G711 U Law or G711 A Law

Configure Microsoft NetMeeting for Voice Over IP

To configure NetMeeting to work with Voice over IP, perform the following steps in the order given:

1. From the Tools menu in the NetMeeting application, select **Options**. NetMeeting will display the Options dialog box.
2. Click the **Audio** tab.
3. Click the **Calling a telephone using NetMeeting** check box.
4. Enter the IP address of the Cisco AS5300 in the IP address field.
5. Under General, click **Advanced**.
6. Click the **Manually configured compression settings** check box.
7. Select the CODEC value **CCITT ulaw 8000Hz**.
8. Click the **Up** button until this CODEC value is at the top of the list.
9. Click **OK** to exit.

Initiate a Call Using Microsoft NetMeeting

To initiate a call using Microsoft NetMeeting, perform the following steps in the order given:

1. Click the **Call** icon from the NetMeeting application. Microsoft NetMeeting will open the call dialog box.
2. From the Call dialog box, select **call using H.323 gateway**.
3. Enter the telephone number in the Address field.
4. Click **Call** to initiate a call to the Cisco AS5300 from Microsoft NetMeeting.

VFC Management

VFCs come with a single bundled image of VCWare stored in VFC Flash memory. Table 3 shows the extension types defined for these embedded firmware files.

Table 3 VFC Firmware Extensions

Firmware	Filenames	Description
VCWare	vcw-vfc-*	Latest version of VCWare stores in Flash memory, including: • Datapath engine • Message dispatcher • DSP manager • VC manager • Process scheduler
DSPWare	btl-vfc-*	DSP bootloader
	cor-vfc-*	Core operating system and initialization
	bas-vfc-*	Base voice
	cdc-*-*	Voice CODEC files
	fax-vfc-*	Fax relay files

DSPWare is stored as a compressed file within VCWare; you must unbundle VCWare to install DSPWare into Flash memory. During the unbundling process, two default lists (the default file list and the capability list) are automatically created, populated with default files from that version of VCWare, and stored in VFC Flash memory. The default file list contains the filenames indicating which files are initially loaded into DSP upon bootup. The capability list defines the set of CODECs that can be negotiated for a voice call.

VFC management enables you to add versions of VCWare to Flash memory (download and unbundle files), erase files contained in Flash memory, add files to the default file list and capability list, and delete files from the default file lists and capability lists.

This section describes the following topics:

- Download VCWare
- Copy Flash Files to the VFC
- Unbundle VCWare
- Add Files to the Default File List
- Add CODECs to the Capability List
- Delete Files from VFC Flash Memory
- Erase the VFC Flash Memory

Download VCWare

To download software to your VFC, you need to do the following:

- Determine that the version of VFC ROM Monitor software is compatible with your installed Cisco IOS image. VFC ROM version 1.2 requires Cisco IOS image 0.14.1 (1.6 NA1) or later. VFC ROM Monitor version 1.2 can be made to work with Cisco IOS image 0.13 (or later) by appending the suffix ".VCW" to the VCWare image stored in VFC Flash memory.
- Determine whether the VFC is in VCWare mode or ROM Monitor mode. The mode, whether VCWare or ROM Monitor, determines how you download software to the VFC.
- Download the software using the appropriate procedure.

Determine the Number of VFCs

To determine the number of installed VFCs and their location, use the following commands in privileged EXEC mode:

Command	Purpose
show vfc *slot* **directory**	Determines the number of installed VFCs and their location.

For each VFC identified and located, perform the tasks described described in the following sections to upgrade system software on that VFC.

Identify the VFC Mode

To identify the mode (whether VCWare or ROM Monitor), use the following commands in privileged EXEC mode:

Command	Purpose
show vfc *slot* **board**	Determines whether you VFC is operating in VCWare mode or ROM Monitor mode.

If the mode is VCWARE, the VFC status will be "VCWARE running." If the mode is ROM Monitor, the VFC status will be "ROMMON."

Download Software (VCWare Mode)

To download VFC software to the VFC while the VFC is in VCWare mode, use the following commands beginning in privileged EXEC mode:

Step	Command	Purpose
1	**erase vfc** *slot*	Erases the Flash memory.
2	**show vfc** *slot* **directory**	Verifies that the VFC Flash memory is indeed empty.
3	**copy tftp vfc** **copy flash vfc**	Downloads the VCWare from a TFTPBoot server into VFC Flash memory or Downloads the VCWare from the VFC motherboard into VFC Flash memory.
4	**clear vfc** *slot*	Reboots the VFC.
5	**show vfc** *slot* **board**	Checks to see if the VFC is back up in VCWare mode.
6	**show vfc** *slot* **directory**	Verifies that VCWare is in the VFC Flash.
7	**unbundle vfc** *slot*	Unbundles the DSPWare from the VCWare and configures the default file list and the capability list.
8	**show vfc** *slot* **directory**	Verifies that the DSPWare has been unbundled.
9	**show vfc** *slot* **default-list**	Verifies that the default file list has been populated.
10	**show vfc** *slot* **cap-list**	Verifies that the capability list has been populated.

After you have completed the preceding tasks, reboot the Cisco AS5300 for these changes to take effect.

Note If the VFC ROM is version 1.1, the image name must end in ".VCW." If the VFC ROM is version 1.2, the image name must start with "vcv-."

Download Software (ROM Monitor Mode)

To download VFC software to the VFC while the VFC is in ROM Monitor mode, perform the following tasks, beginning in privileged EXEC mode:

Step	Command	Purpose
1	**clear vfc** *slot* **purge**	Erase the VFC Flash memory.
2	**copy tftp vfc** or **copy flash vfc**	Download the VCWare from a TFTP server into VFC Flash memory Download the VCWare from the VFC motherboard into VFC Flash memory.
3	**clear vfc** *slot*	Reboot the VFC.
4	**show vfc** *slot* **board**	Check to see if the VFC is back up in VCWare mode.
5	**show vfc** *slot* **directory**	Verify that VCWare is in the VFC Flash.
6	**unbundle vfc** *slot*	Unbundle the DSPWare from the VCWare and configure the default file list and the capability list.
7	**show vfc** *slot* **directory**	Verify that the DSPWare has been unbundled.
8	**show vfc** *slot* **default-list**	Verify that the default file list has been populated.
9	**show vfc** *slot* **cap-list**	Verify that the capability list has been populated.

After you have completed the preceding tasks, reboot the Cisco AS5300 for these changes to take effect.

Note The image name must start with "vcw-."

Copy Flash Files to the VFC

As mentioned, each VFC comes with a single bundled image of VCWare stored in Flash memory. Voice over IP for the AS5300 offers two different ways to copy new versions of VCWare to the VFC Flash memory: either by downloading the image from the AS5300 motherboard or by downloading the VCWare from a TFTP server.

Download VCWare to the VFC from the AS5300 Motherboard

To download the VCWare file from the AS5300 motherboard to VFC Flash memory, use the following command in privileged EXEC mode:

Command	Purpose
copy flash vfc	Downloads (copies) the Flash file from the AS5300 motherboard to the Flash memory on the VFC.

Download VCWare to the VFC from a TFTP Server

To download the latest version of VCWare from a TFTP server, make sure that the file is stored on the TFTP server. If you have a copy of the current version of VCWare on disk, you must store that image on a TFTP server before you can download the file to VFC memory.

To copy the Flash file from a TFTP server, use the following command in privileged EXEC mode:

Command	Purpose
copy tftp vfc	Downloads (copies) the Flash file from a TFTP server to the Flash memory on the VFC.

Unbundle VCWare

VCWare needs to be unbundled for DSPWare to be loaded in Flash memory and the two necessary default lists (default file list and capability list) created and populated with the appropriate default files for that version of DSPWare. Table 4 shows the files associated with each firmware file.

Table 4 VFC Firmware Filenames

Firmware	Filenames
VCWare	vcw-vfc-mz.0.15.bin
DSPWare Initialization and Static Files	btl-vfc-1.0.14.0.bin cor-vfc-1.0.14.0.bin jbc-vfc-1.0.14.0.bin
DSPWare Overlay Files	bas-vfc-1.0.14.0.bin cdc-g711-1.0.14.0.bin cdc-g729-1.0.14.0.bin fax-vfc-1.0.14.0.bin

To unbundle the current running image of VCWare, use the following command in privileged EXEC mode:

Command	Purpose
unbundle vfc *slot*	Unbundles the current image of VCWare.

Add Files to the Default File List

When you unbundle VCWare, the default file list is automatically created and populated with the default files for that version of VCWare. The default file list indicates which files are initially loaded into DSP upon bootup. The following example shows you the output from the **show vfc def** command, which displays the contents of the default file list:

```
router#show vfc 1 def

Default List for VFC in slot 1:
1. btl-vfc-1.0.13.0.bin
2. cor-vfc-1.0.1.bin
3. bas-vfc-1.0.1.bin
4. cdc-g729-1.0.1.bin
5. fax-vfc-1.0.1.bin
6. jbc-vfc-1.0.13.0.bin
```

Under most circumstances, these default files should be sufficient. If you need to, you can add an additional file (from those stored in VFC Flash memory) to the default file list or replace an existing file from the default file list. When you add a specific file to the default file list, it replaces the existing default for that extension type.

To select a file to be added to the default file list, use the following command in global configuration mode:

Command	Purpose
default-file vfc	Selects a file stored in the Flash memory to be added to the default file list.

Add CODECs to the Capability List

The capability list defines the set of CODECs that can be negotiated for a voice call. Like the default file list, the capability list is created and populated when VCWare is unbundled and DSPWare added to VFC Flash memory. The following example shows you the output from the **show vfc cap** command, which displays the contents of the capability list:

```
router#show vfc 1 cap

Capability List for VFC in slot 1:
1. fax-vfc-1.0.1.bin
2. bas-vfc-1.0.1.bin
3. cdc-g729-1.0.1.bin
4. cdc-g711-1.0.1.bin
5. cdc-g726-1.0.1.bin
6. cdc-g728-1.0.1.bin
7. cdc-gsmfr-1.0.1.bin
```

VFC management lets you add additional CODEC files to the capability list to meet the needs of your specific telephony network.

Note The capability list does not indicate CODEC preference; it simply reports the CODECs that are available. The session application decides which CODEC to use.

To add a CODEC overlay file to the capability list, use the following command in global configuration mode:

Command	Purpose
cap-list *file-name* **vfc** *slot-number*	Selects a codec overlay file to be added to the capability list.

Delete Files from VFC Flash Memory

In some instances, you might need to delete a file from the default file list or the capability list or you might need to revert to a previous version of VCWare stored in Flash memory. To delete a file, you must identify and delete the file from VFC Flash memory. Deleting a file from Flash memory removes the file from the default file list and capability list (if the deleted file is included on those lists).

To delete a file from VFC Flash memory, use the following command in privileged EXEC mode:

Command	Purpose
delete *file-name* **vfc** *slot*	Deletes a specific file from the Flash memory on the VFC.

Erase the VFC Flash Memory

When you upgrade to a later version of VCWare, the new files are stored in VFC Flash, along with those already stored in VFC Flash memory—the new files do not overwrite existing files. Consequently, you will eventually need to erase the contents of VFC Flash memory to free VFC Flash memory space. Erasing VFC Flash memory removes the entire contents stored in Flash memory, including the default file list and the capability list.

To erase the Flash memory of a specific VFC, use the following command in privileged EXEC mode:

Command	Purpose
erase vfc *slot*	Erases the Flash memory on the VFC.

For more information about VFC management commands, refer to the “Command Reference” section of this document.

Declarations, Notices, and Network-Related Comments

Notice to Customers In certain countries, use of these products or provision of voice telephony over the Internet may be prohibited and/or subject to laws, regulations, or licenses, including requirements applicable to the use of the products under telecommunications and other laws and regulations; customer must comply with all such applicable laws in the country(ies) where customer intends to use the product.

Configuration Examples

This section provides sample configurations for the following scenarios:

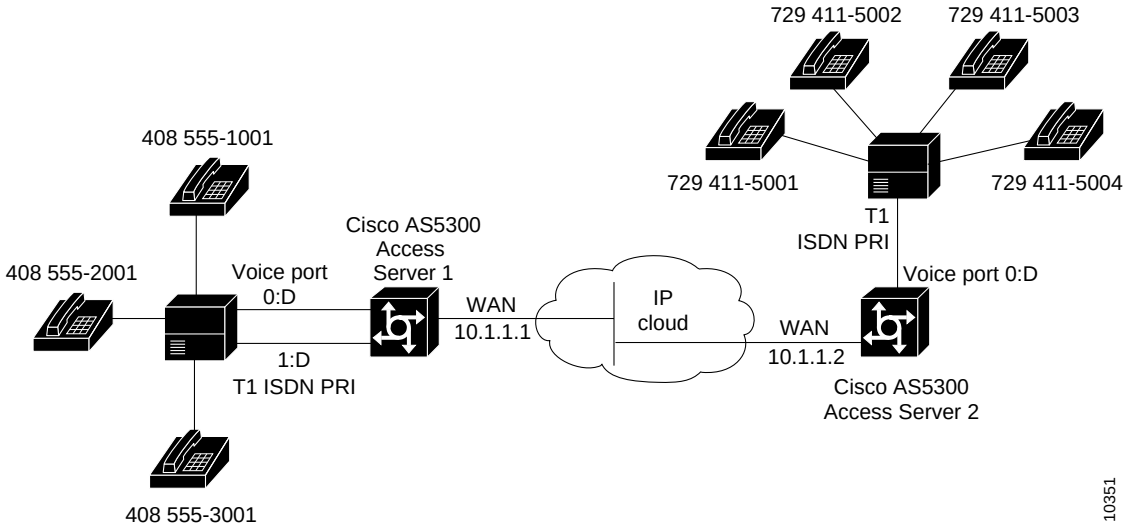
- Linking PBX Users to a T1 ISDN PRI Interface Example
- Configuring Voice over IP for E1 R2 Signalling Example
- Configuring Voice over IP for T1-CAS Example

These configuration examples should give you a starting point in your configuration process. The actual Voice over IP configuration procedure you complete depends on the topology of your voice network. These configuration examples need to be customized to reflect your network topology.

Linking PBX Users to a T1 ISDN PRI Interface Example

This example describes how to configure Voice over IP to link PBX users with T1 channels configured for ISDN PRI signalling. In this example, the company has already established a working IP connection between its two remote offices, one in San Jose, California and the other in Research Triangle Park (RTP), North Carolina. Figure 11 illustrates the topology of this example.

Figure 11 Linking PBX Users to a T1 ISDN PRI Interface Example



Each office has an internal telephone network using PBX, connected to the voice network by T1 interfaces. The San Jose office, located to the left of the IP cloud, has two T1 connections; the RTP office, located to the right of the IP cloud, has only one. Both offices are using PRI signalling for the T1 connections.

To reach a destination in RTP, users in San Jose pick up the handset, hear a primary dial tone, then dial 9, 411, and the destination extension number. To reach a destination in San Jose, users in RTP pick up the handset, hear a primary dial tone, then dial 4. After dialing 4, users hear a secondary dial tone. The users then dial 555, and the extension number.

Configuration for San Jose Access Server

The first part of this configuration example defines dial-in access, including configuring the T1 lines and the ISDN D-channel parameters. For more information about configuring ISDN PRI, refer to the "Configuring Channelized E1 and Channelized T1" chapter in the Cisco IOS Release 12.0 *Dial Solutions Configuration Guide*.

```
hostname sanjose
!
! Define the telephone company's switch type
isdn switch-type primary-5ess
!
! Configure T1 PRI for line 1
controller T1 0
 framing esf
 clock source line primary
 linecode b8zs
 pri-group timeslots 1-24
!
! Configure T1 PRI for line 2
controller T1 1
 framing esf
 clock source line secondary
 linecode b8zs
 pri-group timeslots 1-24
!
! Configure the ISDN D channel for each ISDN PRI line
! Serial interface 0:23 is the D channel for controller T1 0
!
interface Serial0:23
 isdn incoming-voice modem
!
! Serial interface 1:23 is the D channel for controller T1 1
interface Serial1:23
 isdn incoming-voice modem
```

The next part of this example configures number expansion:

```
! Configure number expansion.
num-exp 555.... 1408555....
num-exp 4115... 17294115...
```

The next part of this example configures the POTS and VoIP dial peers:

```
! Configure POTS dial peer 1 using the first T1
dial-peer voice 1 pots
 prefix 6
 dest-pat 1408555....
 port 0:D
!
! Configure POTS dial-peer 2 using the first T1
dial-peer voice 2 pots
 prefix 7
 dest-pat 1408555....
 port 0:D
!
```

```
! Configure POTS dial-peer 3 using the second T1
dial-peer voice 3 pots
 prefix 5
 dest-pat 1408555....
 port 1:D
!
! Configure VoIP dial-peer 4
dial-peer voice 4 voip
 dest-pat 17294115...
 session-target ipv4:10.1.1.2
```

Configuration for RTP Access Server

The first part of this configuration example defines dial-in access, including configuring the T1 line and the ISDN D-channel parameters. For more information about configuring ISDN PRI, refer to the "Configuring Channelized E1 and Channelized T1" chapter in the Cisco IOS Release 12.0 *Dial Solutions Configuration Guide*.

```
hostname rtp

! Define the telephone company's switch type
isdn switch-type primary-5ess

! Configure T1 PRI for line 1
controller T1 0
 framing esf
 clock source line primary
 linecode b8zs
 pri-group timeslots 1-24
!
! Configure the ISDN D channel for ISDN PRI line 1
! Serial interface 0:23 is the D channel for controller T1 0
interface Serial0:23
 ip address 7.1.1.10 255.255.255.0
 encapsulation ppp
 isdn incoming-voice modem
 dialer-group 1
 ppp authentication chap
```

The next part of this example configures number expansion:

```
! Configure number expansion.
num-exp 555.... 1408555....
num-exp 4115... 17294115...
```

The next part of this configuration example defines the POTS and VoIP peers:

```
! Configure POTS dial-peer 1
dial-peer voice 1 pots
 dest-pat 17294115...
 port 0:D
!
! Configure VoIP dial-peer 5
dial-peer voice 4 voip
 dest-pat 1408555....
 session-target ipv4:10.1.1.1
```

Configuring Voice over IP for E1 R2 Signalling Example

The following example configures R2 signalling and customizes R2 parameters on controller E1 2 of a Cisco AS5300. In most cases, the same R2 signalling type is configured on each E1 controller.

```
! Specify the E1 controller that you want to configure with R2 signalling. A controller
! informs the access server how to distribute or provision individual timeslots for a
! connected channelized E1 line. You must configure one E1 controller for each E1 line.
! Configure channel associated signalling. The signalling type forwarded by the
! connecting telco switch must match the signalling configured on the Cisco AS5300.
! The country code is ITU by default.
!
controller E1 0
 framing NO-CRC4
 cas-group 0 timeslots 1-15,17-31 type r2-digital r2-compelled ani
 cas-custom 0
!
controller E1 1
 framing NO-CRC4
 clock source line primary
 cas-group 0 timeslots 1-15,17-31 type r2-digital r2-compelled
!
! Customize some of the E1 R2 signalling parameters with the cas-custom channel
! controller configuration command. This example specifies the default R2 settings for
! Brazil.
!
 cas-custom 0
  country brazil use-defaults
  metering
  category 2
  answer-signal group-b 1
!
controller E1 2
!
controller E1 3
!
! Configure voice port parameters. Be sure that the cptone command value is compatable
! with the country code defined by the cas-custom command. In this example, because
! ITU has no specific cptone value defined and uses aLaw E1 R2 signalling, the GB
! cptone command value is used.
!
voice-port 0:0
 cptone GB
!
voice-port 1:0
 cptone BR
 description Brasil Tone
!
! Define the parameters associated with the VoIP dial peer.
!
dial-peer voice 101 voip
 destination-pattern +500..
 session target ipv4:172.14.25.1
!
! Define the parameters associated POTS dial peer.
!
dial-peer voice 8221 pots
 destination-pattern 011822...
 direct-inward-dial
 port 0:0
!
! Configure LAN interfaces.
!
interface Ethernet0
 ip address 172.13.103.33 255.255.0.0
```

```
 no ip directed-broadcast
 no ip mroute-cache
 load-interval 30
 no cdp enable
!
interface FastEthernet0
 ip address 173.14.25.100 255.255.0.0
 no ip directed-broadcast
 bandwidth 1000000
 load-interval 30
 duplex full
 hold-queue 75 in
!
no ip classless
ip route 223.255.254.253 255.255.255.255 Ethernet0
!
!
line con 0
 exec-timeout 0 0
 logging synchronous level all
 transport input none
 escape-character BREAK
line aux 0
 rotary 1
 transport preferred none
 transport input all
 flowcontrol hardware
line vty 0 4
 exec-timeout 60 0
 password lab
 login
!
end
```

Note Cisco strongly recommends that you specify your country's default settings. To display a list of supported countries, enter the **country ?** command under the **cas-custom** command. The default setting for all countries is ITU.

Configuring Voice over IP for T1-CAS Example

The following example configures T1 CAS parameters on a Cisco AS5300:

```
! Enter global configuration mode.
config terminal
! Enter controller configuration mode to configure your controller port. The controller
ports are labeled 0 through 3 on the Quad T1/PRI and E1/PRI cards.
 controller t1 0
! Enter your telco's framing type.
  framing esf
! Enter the clock source for the line. Configure other lines as clock source secondary
! or internal. Note that only one PRI can be clock source primary and one PRI can be
! clock source secondary
  clock source line primary
! Enter your telco's line code type.
  linecode b8zs
! Configure all channels for E&M, FXS, and SAS analog signalling. Enter 1-24 for T1.
! If E1, enter 1-31.
! Signalling types include e&m-fgb, e&m-fgd, e&m-immediate-start, fxs-ground-start,
! fxs-loop-start, sas-ground-start, and sas-loop-start.
! You must use the same type of signalling that your central office uses.
! For E1 using the Anadigicom converter, use cas e&m-fgb signalling.
```

```
  cas-group 1 timeslots 1-24  type e&m-fgb dtmf dnis
! Configure each additional controller (there are four). In this example, the
! controller number is 1, instead of 0. The clock source is secondary, instead of
! primary. The cas-group is 2, instead of 1
 controller t1 1
  framing esf
  linecode b8Zs
  clock source line secondary
  cas-group 2 timeslots 1-24 type e&m-fgb
! Configure each additional controller.
 controller T1 2
  clock source internal
  cas-group 0 timeslots 1-24 type e&m-fgd mf ani-dnis
 controller T1 3
  clock source internal
! Enter the dial peer configuration mode to configure a POTS peer.
! Specify destination pattern for this POTS peer.
 dial-peer voice 3070 pots
  destination-pattern +30...
  port 0:1
  prefix 30
! Specify destination pattern, and direct inward dial for each POTS peer.
 dial-peer voice 4080 pots
  destination-pattern +40...
  direct-inward-dial
  port 1:2
  prefix 40
! Specify the destination pattern and the direct inward dial for the dial peer.
 dial-peer voice 1050 pots
  destination-pattern +10...
  direct-inward-dial
  prefix 50
! Specify the destination pattern and the direct inward dial for the dial peer.
 dial-peer voice 2060 pots
  destination-pattern +20...
  direct-inward-dial
  prefix 60
 dial-peer voice 5050 voip
  answer-address 10...
  destination-pattern +50...
  end
 end
```